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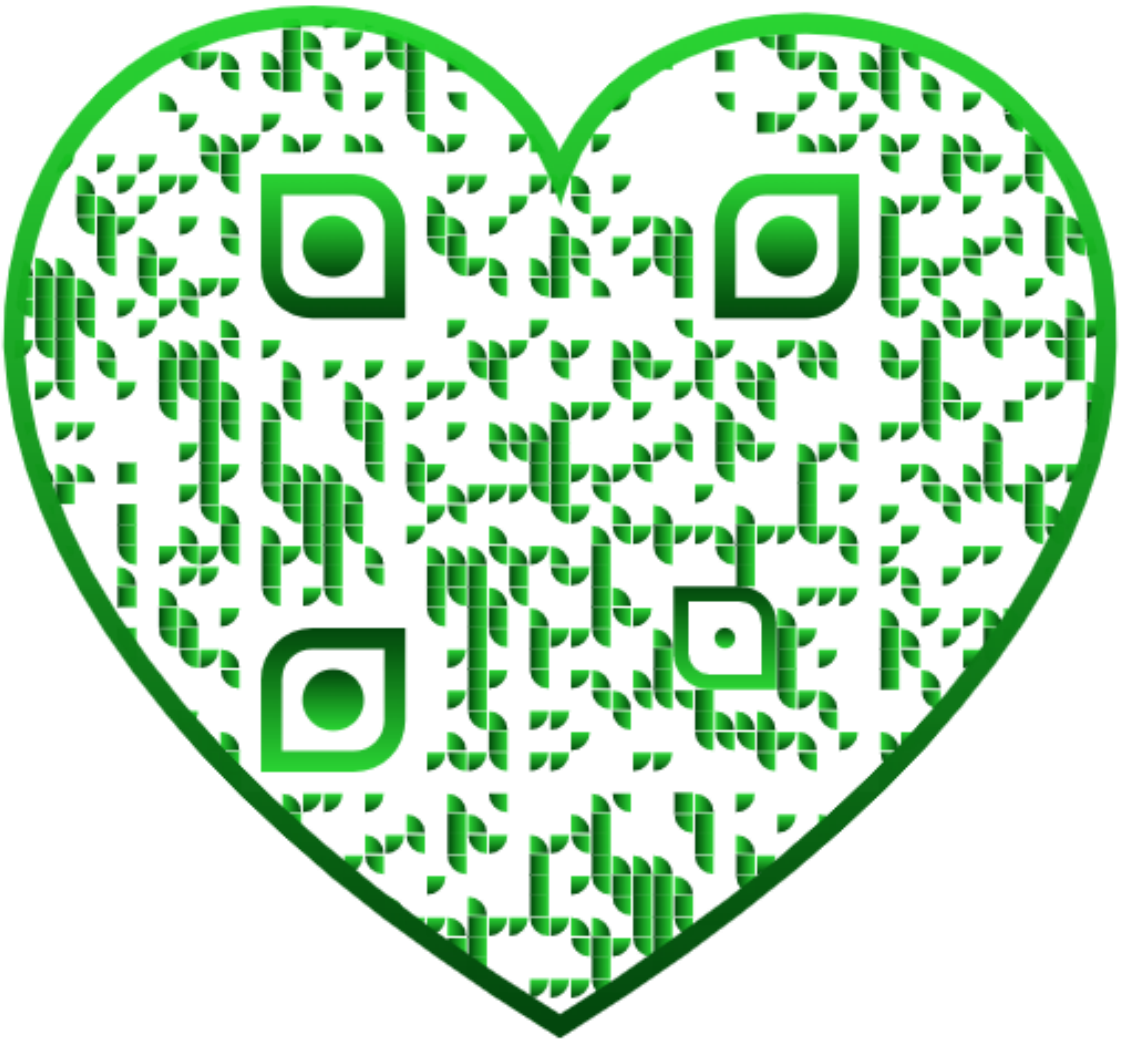

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BEYOND OUR YEARS

It's always uncomfortable when someone says the quiet part out loud. Doing just that at the recent NAMM Show were hardware manufacturers Audioscape, whose Trainwrecked series is, in short, a collection of distressed or 'reliced' outboard equipment.

Now I don't know about you, but I long ago gave up the idea that functionality was the most important aspect of guitar design. If what we really wanted was the most efficient, versatile tool for the job, we'd all be playing Yamaha Pacificas. In real life, appearances do matter, and so the popularity of relic Strats and Teles isn't hard to understand.

But does the same really hold true for studio equipment? Are we in fact so shallow that we're more likely to patch in an LA-2A clone if it looks like it was rescued from a house fire? Surely we are all golden-eared professionals, who make our equipment choices strictly on the basis of sonic integrity? Well, probably not, and the introduction of 'reliced' audio equipment is disconcerting because it pulls aside the curtain and exposes this illusion for what it is.

We are all influenced by appearances. We expect gear to sound a certain way because it looks a certain way, and manufacturers can play on these associations to make their gear more appealing. Nothing made in 2024 really needs to have Bakelite knobs, or Hammertone finishes, or magic-eye meters — but they at least have a veneer of utility. What shocks about the idea of relicing is that there's no conceivable functional reason for it.

But does that actually matter? Studios are environments designed to foster creativity. If we choose distressed furniture for the lounge, or retro lampshades for the kitchen, is that so different from buying gear that has been artificially aged? It all goes to creating a vibe, after all.

In the end, I think, what bothers me isn't that relicing is nakedly aesthetic. It's that the aesthetic misses the point. Fine wine and expensive cheese get better as they mature. Debatably, so do some guitars. But the idea that audio equipment somehow improves with age is just daft. Inasmuch as I love the aesthetic of vintage gear, that has nothing to do with its being scratched and dirty. (I wouldn't want to work in a studio that allowed its gear to get into that state, anyway.) It's because it reflects the highest engineering standards of its time. It reminds us of the creativity and ambition of designers who were pushing boundaries. It's because, although it's now part of our collective past, it once pointed towards the future.

I wonder what equipment made today will inspire similar feelings in 50 years' time.

Sam Inglis
Editor In Chief

"Are we in fact so shallow that we're more likely to patch in an LA-2A clone if it looks like it was rescued from a house fire?"

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dSONIQ Realphones 2

Headphone Correction & Virtual Studio Software

Realphones' sophisticated headphone tuning is now paired with an ever-growing array of virtual monitoring environments.

SAM INGLIS

There are two main strands to dSONIQ's Realphones. It applies measurement-based frequency correction to headphones, and offers binaural simulation of real-world listening environments. Since I first covered it back in *SOS* July 2020, it has continued to evolve. The 1.7 update, reviewed in August 2021, focused on the user experience, introducing an Easy Mode which hid obscure parameters such as headphone Pressure and HRTF Angle. It's now reached version 2, and the main area of development has been the virtual monitoring side of things.

Realphones is installed on macOS or Windows both in native plug-in formats and as a 'systemwide' application. As before, there are three pricing tiers depending on how many headphone profiles and virtual monitoring environments you want access to. The cheapest Standard edition only supports Easy mode, thus omitting most of the detailed control options.

Version 2 has brought with it a change of colour scheme, but the user interface is otherwise broadly the same. That includes Easy Mode, as well as the slightly confusing dual system of conventional presets and Snapshots within Emulation Sets. This would perhaps benefit from being consolidated into a single framework, but it's easy enough to work with once you've located the Lock Parameters options that prevent headphone type and output volume being reset each time you load a Snapshot.

The library of headphone measurements has grown steadily since v1, and in the Professional and Ultimate editions, it's now possible to roll your own, or add custom 'tuning' comprising up to 14 EQ bands. You can also send your personal headphones in for individual measurement, and if you have Sonarworks Reference or Toneboosters Morphit, you can use a Recalibrate profile to 'sniff' headphone profiles from those programs.

In v1, the virtual spaces available were a single control room, a nightclub and a car interior. These have now been joined, depending on the edition, by a whole host of other speakers and rooms, and dSONIQ say they've substantially improved the realism of all the virtualisations. Disregarding the older ones, which are still included as legacy options, there are now around 50 speaker-plus-room combinations in the full set. That includes four 'pro' studios with near, mid and far-field options and more, plus a home studio, a concert hall, nightclubs and a restaurant, and a large number of domestic playback systems: mobile phones, TVs, laptops, Bluetooth speakers and cars.

Back In The Room

Where it's possible to switch between legacy and new virtualisations, there's generally a clear difference, presumably in part because the new measurements were taken at slightly different points within the room or car. I'd be hard put to say which sound 'better' in terms of being more pleasurable to listen to, but the new ones definitely give you a more convincing sense of being present in the space. That sensation is particularly vivid with some of the consumer playback systems, and I several times had the disconcerting feeling of "Oh no, my headphones aren't plugged in and everyone can hear my laptop speakers!"

As with previous versions, I found it well worth experimenting with Advanced Mode parameters such as headphone Presence, room Ambience and HRTF Angle to establish my own preferences. But once you've figured out how you like your virtual rooms served, you can pretty much leave these settings alone and get on with the business of mixing or checking translation. And for both purposes, I think Realphones is up there with the best of its kind.

Obvious competitors that offer similar virtual monitoring and mix-checking features include Sonarworks Reference ID, Embody Immerse Virtual Studio and Slate VSX. VSX is tied to its own headphones, which adds to the cost but allows the software to be optimised for those phones. Of the others, Realphones is the most tweakable, which can be a double-edged sword, but allows you to dial things in exactly as you like. A software-only solution makes a lot of sense for those who already own a decent pair of cans, and Realphones 2.0 is as good as I've heard.

summary

With highly configurable headphone calibration and a broad range of virtual monitoring and mix-checking environments, this is a serious tool for headphone mixing.

$ Standard edition $99; Professional edition $149; Ultimate edition $219.
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— Larry Crane, Tape Op Magazine / Jackpot! Recording Studio

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MATT HOUGHTON

Trident 80B-Pre

Microphone Preamplifier

With separate control over mic and line/instrument inputs as well as variable high- and low-pass filters, Trident's 80B-Pre is more versatile than most 500-series preamps.

Trident's Series 80B consoles, which graced studios from the late '70s onwards, have proven enduringly popular, and their clean-sounding, low-distortion preamps played as big a part in that as the EQ. Now owned by PMI Audio, Trident recently have put a modern version of this preamp into the 500-series format. As I've opined countless times in these pages, most readers are not without clean and capable preamps in their audio interfaces, so why would you splash out another $600 on this one? Well, as with so many outboard preamps, these offer that bit more, in terms of both quality and features.

Overview

The Series 80B consoles had transformer-balanced mic preamps, and this module uses a Lundahl tranformer in that role. The line input is balanced electronically, and the two inputs each have a dedicated gain control: the mic amp ranges from -10 to +65 dB and the line one ±20dB, both being continuous pots. A switch flips between mic and line mode, and when in line mode, a separate high-impedance instrument input on the front takes precedence when a jack is plugged in. This arrangement could prove really convenient in a home studio or songwriter setup, since you could potentially have 'ready to go' gains set for, say, your vocal mic and a bass or guitar, and have the output feed a single input channel on your audio interface. The output signal runs through another Lundahl transformer.

There's switchable 48V phantom power, as well as a polarity inversion switch and an eight-segment LED VU meter to indicate the output level, ranging from -20 to +15 dB. The standout feature for me, though, is a pair of separately engageable high- and low-pass filters, each with continuously variable pots and a generous range (HPF from 30 to 350 Hz and LPF from 2 to 20 kHz and no obvious 'cramping' of the controls. So not only can you remove rumbles or compensate for the proximity effect and roll off the hiss from tape or noisy instruments and effects pedals, but you have the option to bracket any source to tuck it neatly into a mix, at the recording stage or after the fact. If you happen to have a 500-series EQ module that lacks high- or low-pass filters, using this preamp with it will put much more tone control at your finger tips.

Speaking of 500-series EQs, Trident sent the review preamps housed in their Deca Dent 500-series rack, along with pairs of their three EQ modules. We've looked at the Deca Dent and the 80B 50 EQ (https://sosm.ag/trident-decadent-80beq) and the Hi-Lo dynamic filter (https://sosm.ag/trident-hi-lo) previously and they all impressed, but this was the first time I'd tried the doublewide A-Range EQ. I found it a wonderful partner for the 80B preamp. Unlike the 80B EQ, this includes high- and low-pass filters, with a choice of three fixed frequencies. The preamp's filters can usefully augment these, to allow rumble and proximity boost filters simultaneously, for example, or steeper slopes. The four main bands have switched frequencies for easy recall and, unusually, the band gain is set using faders rather than rotary pots, giving a useful visual indication of the setting. Oh... and last but not least, it sounds gorgeously smooth!

Bottom Line

So... is the 80B preamp worth it? It's certainly well built, and it sounds clean yet classy, and would be a great, versatile option for someone looking to build a 500-series channel strip. As mentioned above, the separate line and mic stages and the front-panel DI mean it could work very well for a home-studio or songwriter setup. And having those filters on board could make it indispensable to some engineers, making its price tag very reasonable.

The rack sent for review featured pairs of various Trident 500-series modules, including the 80B preamp and EQ, the Hi-Lo dynamic filter and, on the right, the A-Range EQ.

summary

A nicely thought-out 500-series preamps with some very handy features.

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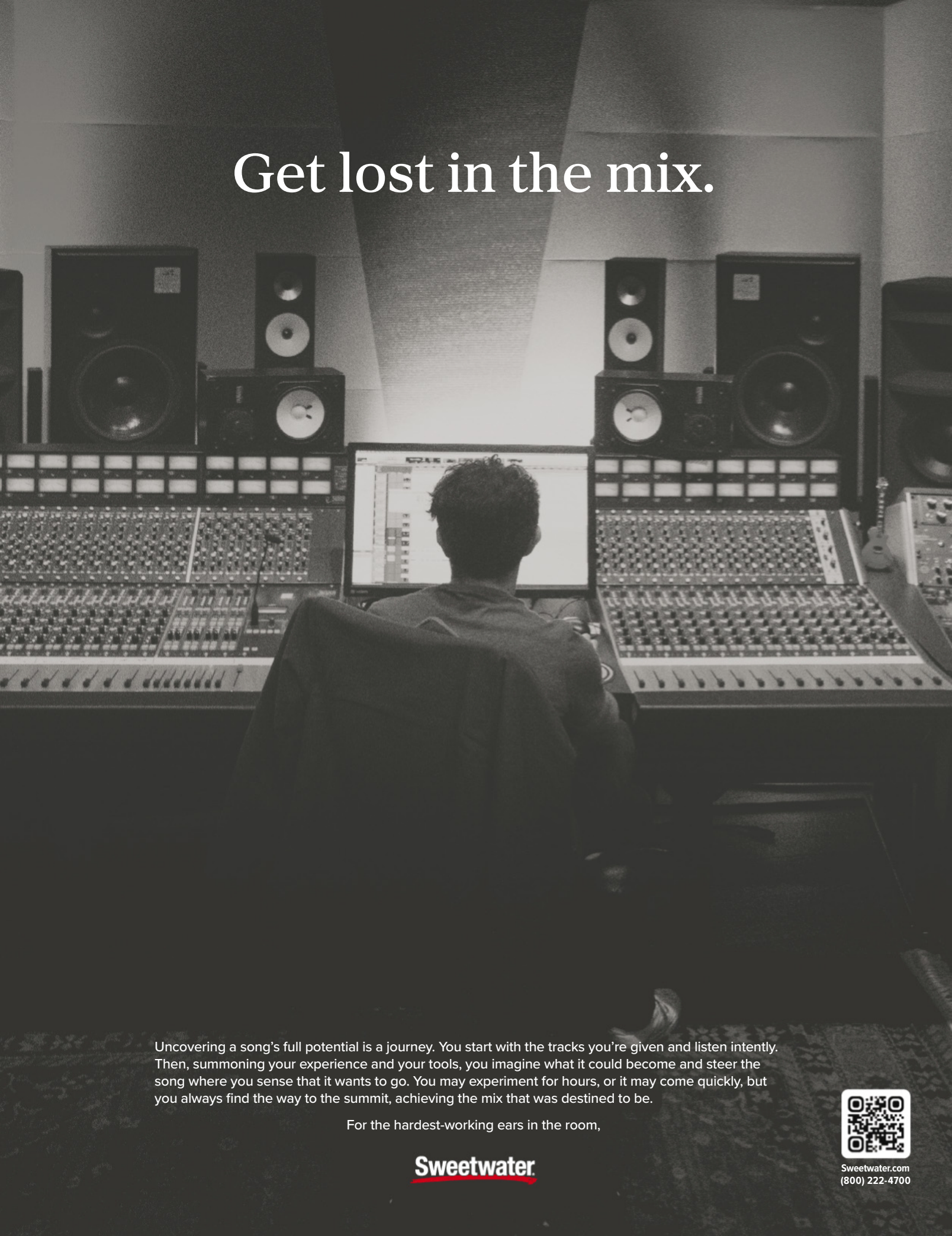
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Sennheiser HD 490 Pro

Open-backed Headphones

Sennheiser prove there's still room for fresh thinking in headphone design!

SAM INGLIS

Like many headphone manufacturers, Sennheiser don't only operate in the studio and pro audio markets. They are also major players in consumer audio and hi-fi, and the streams quite often get crossed. Sennheiser's HD 650 open-backed headphones, for example, are officially located on the hi-fi side of the gulf, but they remain a popular choice for studio applications. On 'our' side of the divide, meanwhile, pro audio staples like the HD 25 MkII are now joined by the new HD 490 Pro. These cost slightly less than the HD 650 and, like that model, are conventional over-ear, open-backed, moving-coil headphones. Unlike the 650, however, they are very much targeted at studio applications. So, although they are far from being the most costly model that Sennheiser make, the HD 490 Pros are in a sense the flagship model of the pro audio line.

It's relatively easy to identify features that mark some headphones out as *not* being designed for pro audio. There's no obvious studio role for wireless Bluetooth connectivity, active noise cancelling or cat ears, for example. But what, apart from a balanced, neutral sound, makes this new set of cans particularly well suited for audio engineering? Well, Sennheiser's engineers have come up with several interesting design features that mark the HD 490 Pros out as different not only from other Sennheiser models, but also from rival products.

Case Study

The HD 490 Pros are available in two versions, the main difference being that the Plus edition comes with a very nice fitted case. Given the relatively modest price differential, I think it's definitely worth spending the extra if they're ever likely to come on the road with you. Plus purchasers also get a 3m cable as well as the default 1.8m one, and a spare for the removable foam pad that cushions the headband against the top of your head.

Talking of cables, one nice and arguably 'professional' feature is that both earcups have a mini-XLR socket, allowing the cable to be plugged into whichever side is most convenient. A small plug closes the socket that is not in use. Not exclusively professional but certainly very welcome is that these XLRs are four-pin sockets, allowing the HD 490 Pro to be driven from balanced headphone amps if you have the correct cable.

Still talking of cables, the supplied mini-XLR to jack wires are interesting in their own right (I can't believe I just said that about a headphone cable). They are, at heart, straight cables, but with a short curly section located a couple of inches below the XLR. This, I assume, is related to one of the patented innovations that Sennheiser claim for the HD 490 Pro, which is a dramatic reduction in cable-borne noise compared to conventional designs. It really works, too: flick the short section of straight cable above the curly segment and you'll hear the usual annoying thumps and clicks. Do the same to the main part of the cable and the only sound you'll hear is what reaches your ears acoustically through the HD 490 Pros' open grilles.

And talking of grilles, Sennheiser also claim a long list of innovations relating to the design of the earcups and the drivers. These include a "geometrically angled earcup design" that "optimises transducer placement for a consistent listening experience"; an "innovative low-frequency cylinder system for full, accurate, and clearly defined low end"; an "open-frame architecture" which "reduces total harmonic distortion and minimises resonance"; and "Braille-guided L/R earcups" for blind and partially sighted users. Their engineers have also, at long last, thought about the comfort of spectacle wearers, with "grooved ear pads for eyeglasses".

Pad Reputation

It is reasonably well known that the geometry and material of headphone earpads can make a significant difference to the sound of a pair of headphones, and there's a thriving community of DIY headphone enthusiasts who like to experiment with different fabrics, fillings and dimensions. (Wait til they hear about Sennheiser's new cable!) Uniquely, at least in my experience, the HD 490 Pros are supplied with not one but two different pairs of pads. The factory-fitted velour earpads can be swapped out for an alternative set of cloth-covered pads. As usual, this is a straightforward process, with the replacement pads simply clipping into place. The two types are, respectively, designated the Producing and the Mixing earpads. Sennheiser publish frequency response plots which show that the velour Producing pads introduce a gentle low-frequency shelving boost, starting at around 400Hz and about 4dB up at 100Hz. There are also differences of detail further up the spectrum, but nothing dramatic.

Sennheiser say the choice is offered in response to one of those awkward dilemmas that plague manufacturers. If you ask audio professionals what they want, they'll tell you they prize a neutral, uncoloured sound; but actual sales figures suggest that 'warm'-sounding headphones are more popular than truly neutral models. »

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Comfort Listening

Most new headphones these days seem to be low-impedance designs. With a nominal impedance of 130Ω, the HD 490 Pro are more so than the 300Ω HD 650, but still higher than most. Still, with a sensitivity of 105dBSPL for a 1Vrms 1kHz input, they shouldn't present too many problems to mobile devices and other low-powered headphone amps. Another intriguing contrast in terms of specifications relates to distortion: the HD 650s achieve a remarkably low THD figure of 0.05%, whereas the HD 490 Pros are merely very good, at 0.2%.

Unusually, one of the HD 490s' most obvious competitors is also made by a company within the Sennheiser group. The Neumann NDH 30s are, likewise, open-backed headphones targeted squarely at pro audio, and although they appear to have nothing in common design-wise, their specified sensitivity and impedance figures are very similar. Once again, distortion is where the two differ, with the Neumann design nearly an order of magnitude better at 0.03% THD.

I don't have a pair of HD 650s, but I was able to compare the review HD 490 Pros directly against the NDH 30s, as well as the Shure SRH1840s, my longstanding reference for neutral sound in open-backed moving-coil headphones. Having briefly tried the velour earpads, I quickly switched to the fabric ones and didn't go back; to my ears, the sound is a hint on the warm side of neutral even with the Mixing pads. Very much like the Neumanns, there's a gentle but noticeable bump somewhere in the 100-150 Hz area, and in this case, it slightly overshadows what might be called the 'true' bass region below 100Hz. It's subtle, though, and you get used to it almost immediately.

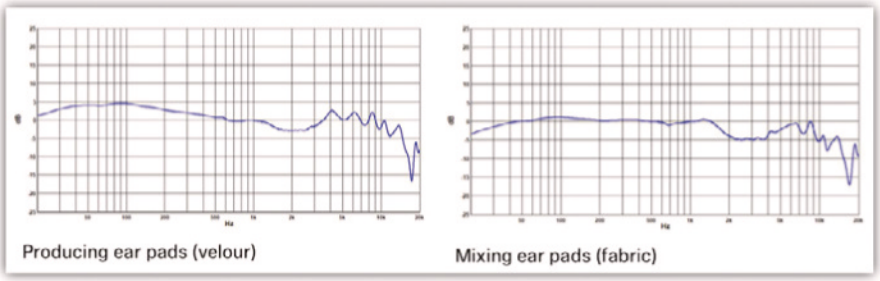
In the NDH 30s, this warmth is paired with a hint of softness in the midrange, most noticeable around 2kHz or so. That's not at all the case with the HD 490 Pros, which to my ears are just on the confident side of neutral when it comes to midrange presentation. It's a voicing that I find much more useful for music mixing and editing than the sparkly treble and exaggerated bass that some headphones deliver. It's also very similar to the sound of the Shures. In fact, I'd say that the SRH1840s subjectively sit in between the Neumann and Sennheiser models from a tonal point of view, but rather closer to the latter. Depending on your preference, I can imagine that any one of the three might strike you as nailing your idea of 'neutral'.

The Plus edition comes with a smart and practical carry case.

And although the HD 490 Pros' distortion performance may be inferior on paper to that of the NDH 30s, it certainly isn't a problem in practice. If distortion is likely to be audible at all, it'll be on music with heavy, low bass, and my journey through the world of Bass Boosted Punjabi Music playlists didn't provoke any misbehaviour that I could detect. 808s go lower on the NDH 30s, though, which definitely have the edge when it comes to representing sub frequencies.

I like the tonality of the HD 490 Pros, then; and I like lots of other things about them, too. I found them much more comfortable to wear than the Neumanns, perhaps even more so than the Shures: maybe that spectacle groove makes a real difference. I also really, really enjoyed listening to them. The HD 490 Pros present music, especially guitar-based music, with a confidence and immediacy that puts a smile on your face. Yet at the same time, they're certainly neutral enough for analytical listening. I must have listened to *Parachute* by the Pretty Things hundreds of times in the last 30 years, but I'd never noticed the strange noise near the start of 'The Good Mr Square' until I heard it through the HD 490s. And from a production point of view, I was particularly impressed with how they put across vocal sibilants. Many headphones exaggerate these or somehow mangle them; the HD 490s made them sound like the natural part of the human voice they are. In fact, they clearly bettered the Shures in this regard — so much so that I subjected them to the ultimate high-frequency torture test, otherwise known as the tambourine at the start of the mono mix of 'Hey Jude'. They passed with flying colours, leaving the Shures well behind and perhaps even beating out the Neumanns.

The bottom line here is that if I had to mix on headphones, I'd be happy doing so with any of these three models. But the HD 490 Pros' comfort, adaptability and very slightly mid-focused tonality are all qualities that I personally find very appealing, and the bonus is that they are the least costly of the three. Especially if you work with the kinds of music that are recorded using microphones rather than made with synths, I think they are very hard to beat at the price. ■■■



The main difference between the Producing and Mixing earpads lies in the low end.

summary

The HD 490 Pros are bristling with innovation, aggressively priced, supremely comfortable — and they sound very, very good.

$ $399 or $479 for Plus edition.
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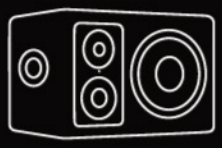

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Undertone Audio UTEQ500

500-series Equaliser

With a chameleon-like ability to replicate the curves of almost any analogue bell or shelf EQ, this has to be one of the most versatile equalisers out there.

MATT HOUGHTON

The UTEQ500 is a three-band EQ, and yes, I know you've all used those before — but not one like this. Because, despite having only six dual-concentric knobs and a few toggle switches to play with, the UTEQ500 is probably the most versatile 'console-style' analogue equaliser I've ever had the pleasure of testing. A solid-state design, this single-slot 500-series module is electronically balanced, and the filtering is performed using resistor-capacitor networks. With no transformers or inductors at play, it won't surprise you to learn that it doesn't sound 'vintage' in terms of harmonics and saturation. Rather, it's all about the curves the way you can manipulate them, though there is still definitely a sweet spot into which you can drive it for a little 'crunchy' attitude if you like to do that.

Basics

All three EQ bands are identical, other than that they operate in different frequency ranges. The frequency is determined using a pot rather than switches, so you can set each band freely within its range. Each band can also be engaged/bypassed independently. The high band, at the top of the module, runs from 1.4 to 22 kHz; the mid band controls are switchable to operate in either a low-mid (90Hz to 1.4 kHz) or high-mid (390Hz to 6kHz) range; and the low band frequency control covers 20-340 Hz. There are also small toggle switches that allow you to set each band to boost, cut or notch — more on that below.

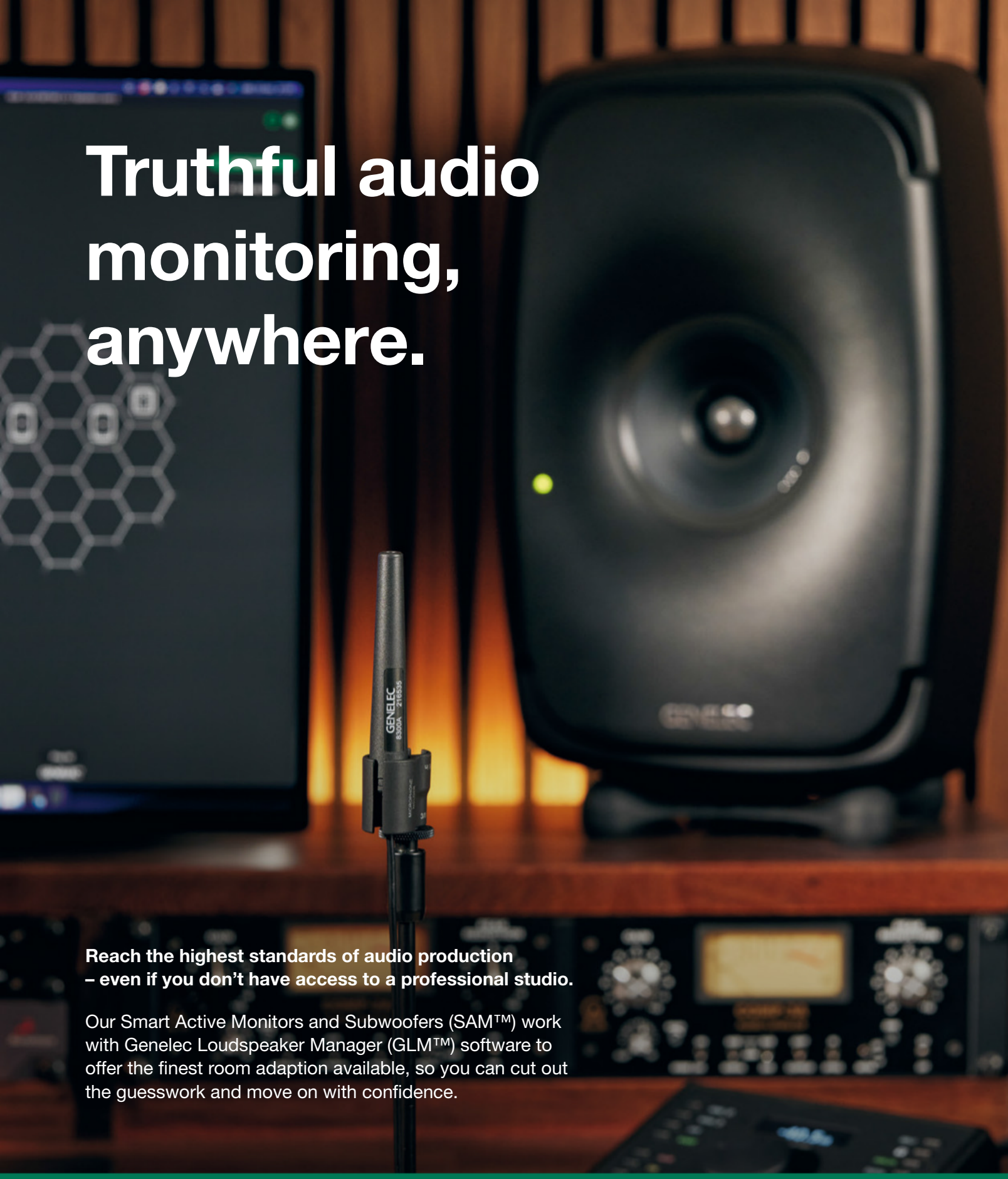
All of the six knobs are dual-concentric types. That's a good call on a 500-series module because, as on a console, panel real estate is a precious commodity, and this choice permits lots of functionality to be accommodated while still allowing access for large fingers and thumbs. Of course, it also means that there's more going on here than first impressions might suggest, and if you're one of those who likes to dive in without reading up on what the controls do, you might like to approach this particular EQ differently.

The outer knob on the left of each band is its gain control. This is marked from 0-10 rather than in dB, presumably so that the legend is uncluttered and relates meaningfully to the inner knob as well as the outer one. The gain pot can be set anywhere from 0-15 dB, so with the boost/cut switch, the total range is ±15dB. When in notch mode, though, it can go deeper still, as I'll discuss below. The outer knob of each band's control on the right is the frequency selector. Then we have the inner controls, with the blue-capped one on the right being a Q (bandwidth) control and the black-capped knob one a Shape control.

At its fully anticlockwise position, the blue Q control gives you its widest bandwidth, with a Q of 0.3 delivering generously wide, gentle curves. At the opposite extreme you get what, for analogue hardware, is a very surgical Q of around 10. So with just the two outer controls and the blue inner one, you already have a pretty conventional three-band analogue parametric bell EQ.

Shape Of U

To use it in that way, you need to set the black Shape control fully anticlockwise. And it's when you start twisting the Shape knob that things start to get a bit less ordinary. Apparently, this feature started life as an internal control »



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» on a prototype, so that Eric Valentine could decide how he wanted his EQ's shelf to sound — most EQ designers choose what sort of curve they want in a shelf — but he had so much fun tweaking it that he felt it should go on the front panel as a user control.

In broad terms, you can think Shape as a control that morphs the band between a bell and a shelf. Taking the low band as an example, let's assume you've dialled in a boost Shape fully anticlockwise: a plot of the frequency response would show a bell shape, falling off either side of the chosen centre frequency. But as you then start to turn the Shape control clockwise, everything below that band's centre frequency begins to be lifted, while everything to the right remains as it was with the bell boost. Keep turning, past the point at which the curve resembles a conventional shelf, and you start to modify the shape of that shelf, with the selected frequency dropping while the shelving boost remains. Take it further towards fully clockwise, and you'll introduce a small dip just above the selected frequency, after which the curve rises to give you your shelf — a shape that's reminiscent of some classic vintage EQs, but here it's much more controllable. Finally, throw in the Q setting, which remains active for both bell and shelf alike, and the three bands, and you can achieve some pretty radical curves indeed. For instance, with narrow bandwidths you can get huge, narrow boosts and cuts either side of the selected frequency, and very steep slopes on your shelves. Once you know how to drive this thing, you can do some pretty cool things to sculpt the low end with just a single band, and that still leaves you with two more bands to throw at the signal. So it's powerful, hugely versatile stuff.

If you want tangible examples of just how versatile it is, it's well worth checking out Eric Valentine's 'deep dive' video on Undertone's YouTube channel. In this, he demonstrates the UTEQ500's ability to almost perfectly match the curves of the Pul-Tec, Neve and API EQs that he'd used on a demo mix. There are *some* differences when looked at on an analyser, but they're vanishingly small. (While there are sonic differences, it also goes to show just how much the EQ curve contributes relative to the harmonic distortion 'character' of an EQ).

Curve Learning Curve

As part of my review tests, I attempted to do something similar and, working with an analyser to visualise and compare the curves, I was able to match some of my own hardware EQ settings pretty closely. It has to be said, though, that when 'flying blind' and relying exclusively on my ears, the different control sets of different EQs always tend to lead me toward slightly different (albeit equally valid) decisions, and this EQ is no exception to that. As I worked with the UTEQ500 on more material, I soon got into the habit of dialling in a pretty basic bell boost/cut with a reasonably wide bandwidth as a starting point, and then manipulating the Shape and Q controls, in that order. I found this a pretty reliable way of getting things where I wanted with the UTEQ500. You might find that a different tactic works better, but the point is that there's a learning curve to this thing: with such radical EQ shapes possible, you'll want to take a while to figure out an approach to using this equaliser that allows you to dial in the sound you want quickly.

Super Notch

You've probably noticed that I've not really said much about the notch filter setting yet. A notch filter is a pretty rare feature on a console-style equaliser, yet the ability to surgically cut out an offending frequency can sometimes be really useful — and with this one, you can cut by over 50dB, pretty much eliminating the selected frequency. The resonance (Q) control is still in play in Notch 'mode' too, allowing you to make the cut as wide or narrow as you want.

While that's all pretty cool, the UTEQ500's notch filter doubles up as an all-pass filter — and that's *really* cool! This is possible because the UTEQ500 is a parallel EQ: the boost mode uses a signal that's the same polarity as the dry signal, so augments it, while cut mode uses a polarity-inverted version of the parallel signal, so subtracts from it. But for the notch, things are configured slightly differently. As you turn the gain knob, you start to bring in the notch filter, and by the time you reach the knob's midway position, the notch is at its deepest. Keep on turning, and the notch recedes until, at the fully clockwise position, it once again has no impact on the frequency response, but the band is still polarity inverted. So when working with two signals, this gives you the ability to massage their phase relationship in a frequency-specific way, allowing you to achieve sounds that you can't simply by 'flipping the polarity' on one or other signal. It's a wonderful tool for working with multi-miked guitar cabs, or when blending more than one amp to achieve the desired tone.

Verdict

This might not be the most affordable 500-series module out there, but I can't think of anything else that does what this does, and you have to be really miserly to find anything to criticise — the best I could come up with is that there's no power-on indicator! So I suppose it's bound to command a premium. Sure, some other EQs will sound a little different from this one, either because they have more bands or do more than just EQ: they may have transformers, inductors and/or valves that impart a certain character, for example. But I'll bet they can't do more in terms of manipulating a signal's frequency and phase. You really can replicate pretty much any curve with just a few controls, and the notch and all-pass filters are the icing on a very special cake. This is one of those ideas that's so simple, yet so powerful that it makes you wonder why it's taken someone so long to come up with it!

As I mentioned above, there is a bit of a learning curve to this thing, but once you have your head around it (watch the aforementioned video and you soon will) then not only does it become really easy to control, but you'll probably find yourself leaning on this EQ more than your others. I should, I suppose, note that the UTEQ500 lacks any means of low- or high-pass filtering a signal — but that's by design, as Undertone plan at least two more 500-series modules, including a mic preamp and an additional EQ with a high- and a low-pass filter plus an additional mid band. Pairing the two EQs would give you a full four-band EQ with high- and low-pass filtering. Again, that combination wouldn't come cheap — though if this is out of your budget, you may like to know that Mr Valentine is rumoured to be working on a plug-in... ■■■

summary

A supremely versatile three-band analogue EQ, with some very clever features — not least the ability to use the notch EQ as an all-pass filter!

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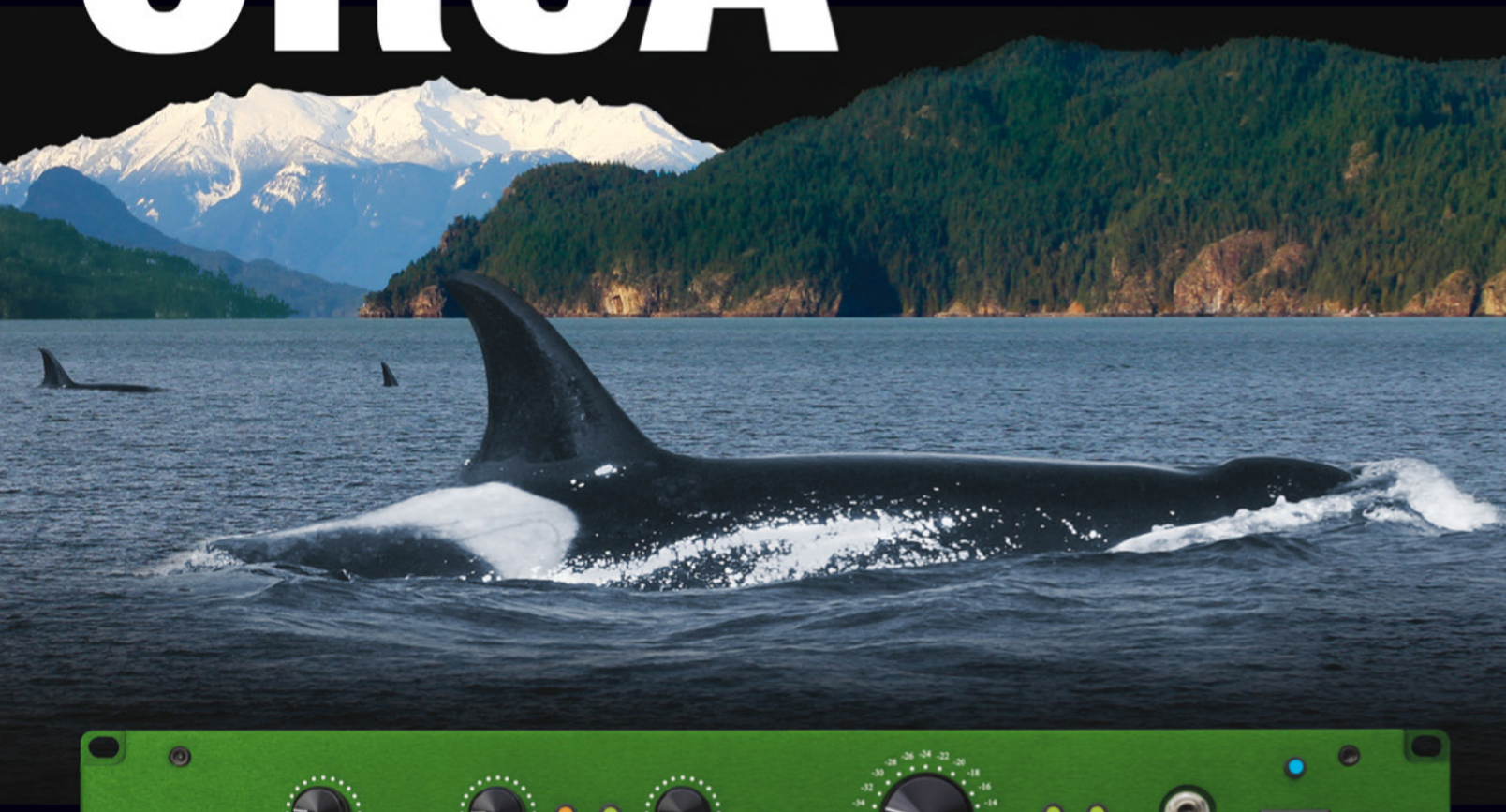

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Solid State Logic B-DYN 500-series Dynamics Processor

Meet the older, hard-hitting version of SSL's 4K channel compressor!

MATT HOUGHTON

SSL's new B-DYN 500-series module is based on the peak-sensing VCA compressor and expander/gate from the 4000B console, an early version of the famous '4K' console released in 1976. While modern components and manufacturing methods are used here, the B-DYN is faithful to the original design, except that there's no output level control that would be a counterpart to the console's channel faders (which operated on the same VCA as the dynamics processor did). There's an automatic gain makeup facility, though.

The compressor has fixed attack (circa 30ms), which tends to let through transients and note onsets but little else before you notice the gain reduction. Consequently, it can sound pretty 'snappy' or 'grabby'. There's a good range of release options, including a programme-dependent Auto setting. The expander (2:1 ratio) and gate (20:1) have a fixed attack, continuous range and threshold controls, and a choice of four release times.

Plan B

With a single B-DYN sent for testing, everything I processed was mono, but two modules in adjacent slots (most 500-series racks support this) can be linked for stereo use. I used it as an 'external plug-in' on a mono bus in a few of my recent mix projects, so I could audition a range of 'real' sources through it: a singer-songwriter track (vocals and a steel-strung acoustic guitar); a more energetic pop-rock track, with singing, bass, lots of guitars and a drum kit; and, finally, a dance track made largely with synths and samples.

Used as a conventional compressor on the acoustic track, the B-Dyn did a serviceable job of containing the vocal, but it wouldn't be my first port of call for gentler parts such as this. There's a handy de-esser setting, though, with very fast (2ms) 10:1 compression reacting to a high-pass filtered (12dB/oct at 7kHz) signal. I'd happily have put this to use here, and it also worked well on the pop-rock track's overheads (albeit made mono) for taming some over-enthusiastic hi-hats. Experimenting with the acoustic guitars, I found the compressor could be used to really emphasise the attack of finger-picked notes or strums, and depending on the part, the gate could shorten notes nicely too — not what I wanted on this project, but you can do a lot to change the feel of a part.

With the two songs that didn't need to sound so 'natural', the B-DYN was a real joy. The more energetically delivered lead vocal of the pop-rock song was controlled assertively, giving it an edge and attitude that brought it right up front. But the B-DYN came into its own when used to reshape individual drums, percussive loops and the drum bus. Bring the gate (20:1 ratio) or expander (2:1) into play alongside the compressor, play with its threshold and release, and you can quickly dial in the hardest, tightest of drum sounds. It's easy to get the pumping effects that can lend a groove some urgency, but the 6dB/octave 185Hz side-chain high-pass filter can counter this too.

It's perhaps worth noting that I had to switch my Ferrofish A32 converter's outputs from their default level setting to their highest, and feed them a pretty loud level signal from the DAW, in order to give me a useful range on the compressor's threshold control. But having done that, it was easy enough to make those LED meters light up. Also note that although the auto gain makeup is broadly helpful, the level can jump a little as you tweak; I'd have liked the option to turn it off, plus a manual output level control. Finally, its sheer punchiness on drums means that this thing is crying out for a wet/dry blend control, so you don't have to rely on a console or your DAW's routing and delay compensation system for parallel processing.

DYN'ing Out

We have an abundance of compressor options today, so I usually look to outboard to do something different. That's the oposite of what people might have looked for in a console's channel compressor 40 years ago, so I can see why SSL changed the design in later versions of the console. While this compressor *can* be used in subtler ways, really it's all about the attitude: snappy, edgy, punchy, grabby, aggressive. Brash and in-your-face. Push the input levels and you're greeted with a harder-edged crunch than the saturation I'd associate with the more relaxed 4000E. But while I do regard the B-DYN as an 'effect compressor', it's also versatile: the gate/expander can be very handy when recording drums, for example, and the compressor doubles as a fast and effective de-esser. All of which makes it decent value for money. ■■■

summary

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CHARLOTTE STERLAND

DJ.Studio Playlisting Software

DJ.Studio is a one-stop shop for creating professional playlists.

DJ.Studio is a virtual DJ editing suite that hands some of the more time-consuming and less glamorous parts of DJ'ing over to the computer. The idea is that automating tempo and key matching gives the DJ extra time to concentrate on the more creative aspects of the job, such as the sound of a transition, overlaying samples and enhancing other mix subtleties.

DJ.Studio also offers YouTube mixing, direct-to-Ableton project exporting and even video editing, not to mention mixing features ranging from filtering to beat matching. The software could also be useful for anyone wanting to put together longer programmes that incorporate different sections of music, other audio and even speech.

Essentials

Easier editing of mixes is at the heart of what DJ.Studio does. The software uses machine learning to analyse a set of tracks selected by the user and arrange them into a suitable order for a set. The process is prompted by clicking on Automix; DJ.Studio will then identify the root note and key of the imported tracks, and make an order suggestion in a matter of seconds.

Software that can identify the tempo, key and other essential features of music is not new, as users of MixMeister (the software that partly inspired DJ.Studio) will know, but there are differences. MixMeister analyses tracks for tempo and key, but does not offer arrangement suggestions.

There are four modes for viewing tracks. In Carousel view, the tracks appear on a horizontal timeline, with overlaid transition edits and a carousel of track artwork above — useful for a visual cue as to where you are in a set. In Playlist view, the tracks are shown vertically in the order they will be played, with tempo, key and various other measures listed alongside. In Studio view the tracks again appear in a horizontal timeline, this time with per-track level, EQ and filter tools. Library view lets you browse the tracks that you've already uploaded in DJ.Studio, and your music library in Rekordbox, Traktor and so on.

Each mode has its benefits. Playlist mode is easier for viewing parameters like tempo alongside the track name, just as in other playlist software, while in Studio mode, volume and other levels can be seen.

Transitions

DJ.Studio's Automix function is used to mix between tracks and offers a number of different transition effects, from straightforward fades to more exotic options. For example, you can use Echo Out to apply a delay effect to the outgoing track, allowing space for the second track to slowly come in. Conversely, Echo In affects the incoming audio in the same way. Bass Swap fades the low end of the outgoing track and the high end of the incoming track, effectively swapping one track's bass for another's. Loop Out will loop a section of the outgoing track, allowing you to extend it while you bring in the next track.

Transitions with effects, as opposed to 'transition effects', are also available, and include a white noise-based riser, downsweep and swoosh, high- and low-pass filters, delay and reverb. These can be applied to the incoming or outgoing tracks and the levels can be automated. »

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While the Automix functionality is impressive, you can also precisely edit changes between tracks with DJ.Studio's Transition Editor. This is where you determine how and when the software mixes between tracks; a number of editable preset transitions are available, to be used or modified as you choose. Basic choices include fade-in, fade-out and crossfade, which are all fairly self-explanatory, and all these basic types can be edited to taste.

Effects, Automation & Tempo Changes

DJ.Studio offers a range of effects that can be applied to tracks and sample tracks (more on the latter in a moment). Currently the effects list includes a three-band EQ, a high- and low-pass filter, echo, flanger, white noise and pitch. These effects can be automated by recording your live tweaking or using DAW-style automation nodes to draw in the information.

Tempo is set in bpm along the length of the mix, and adjusted using value nodes. The tempo can also be slowed down and sped up manually along the length of the track and can be reset back to the original track speed with the Auto key. Drawing out a beat transition over a long time — say, one minute — makes the transition sound smooth and less obvious. Tracks can also be edited so that the tempo is changed but the key remains the same (if, say, the track might need to match another track's speed to make a transition sound slick), so it doesn't sound out of place in the mix as a whole.

Sampling

Two sample tracks appear below the deck tracks when enabled in settings. Any number of sound files can be added to the samples folder, and then dragged to the samples tracks. This is a great way to play with samples over a mix, in a different manner to button-pushing on mix decks. Samples of any length can be used, so a whole track could be mixed alongside one of the main tracks in the decks, effectively giving you a third deck.

Samples can have filters added and their tempo changed, just as for the main tracks; however, all tempos also follow the 'master tempo' for all tracks, keeping the whole mix together. This is confusing if there are suddenly too many things happening, so it may be better to stick to shorter samples, but some rather interesting things can be done with longer files.

Using one of the sample tracks to layer a sample over a transition.

Playlisting

YouTube integration is a great feature of DJ.Studio. The software searches for tracks on YouTube and then makes them available to be mixed into a set. The track remains on YouTube — it's not downloaded — so every time it's played in DJ.Studio it receives a view. Spotify playlists are also supported, and if you upload one to DJ.Studio, the software will search for the corresponding track on YouTube and stream it from there. You can also create mixes with Beatport or Beatsource streaming files: try out any track in your mix, then once you're sure it works well, you can buy the final tracks and record the mix.

While this maybe not an obvious go-to feature for a club DJ, it makes it possible to work with a track on the internet to be sure it fits in your mix before spending any money. Buying track after track can get expensive, and this feature allows you to experiment before parting with any cash. Great for anyone building a new library, or creating a DJ set in a genre they are less familiar with.

The Carousel view shows all the tracks and transitions floating above our decks.

Conclusion

DJ.Studio is a really interesting product for those who want to be able to generate great DJ sets digitally, but who want a program that automates some of the more mundane parts of set mixing. It's a superb tool for generating quick mixes, and also makes it easy to edit them, for those last-minute changes. And it's not just for club DJs: there are obvious applications for radio DJs and producers as well as podcast creators and all audio mixers who want certain aspects automated, so they can focus on creativity in the mix.

DJ.Studio has an Ableton export option, so the project can be exported either as a wave file or an Ableton Live workstation project file with all edits in place. Sets can easily be exported to Mixcloud or YouTube also, or as an MP3 or WAV file; and it's easy to add artwork to that export. Looking forward, DJ.Studio say they are developing partnerships with more software and Internet platforms, which should give musicians even more options for the software they use alongside it, as well as where they mix and buy their audio content from.

It's not DJ.Studio's intention to replace live DJ'ing or hardware mixing, but it's a welcome tool for any DJ or production studio and can create mixes fast, leaving you to concentrate on the creative part.

summary

DJ.Studio provides everything you need to craft sets of songs or sounds with expert detail.

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McDSP BOB APB

Bass Enhancement Plug-in For APB

McDSP's latest hybrid plug-in refreshes the parts of the frequency spectrum other tools can't reach.

SAM INGLIS

As well as being the greenest item in any given studio, McDSP's Analog Processing Box is one of the most versatile. It hosts an array of configurable analogue devices that can implement many different styles of dynamics processing, ranging from emulated variable-mu valve compression to hard limiting, as well as subtle or less subtle saturation. These are bookended by D-A and A-D conversion and accessed through a Thunderbolt connection from your computer, from user interfaces that behave exactly like conventional plug-ins. APB processors 'live' in DAW insert slots and can be stored, recalled, loaded, saved and automated in exactly the same way as any other real-time plug-in, the only caveat being that, for obvious reasons, faster-than-real-time bouncing is not possible.

Say Hello To BOB

Since its launch back in 2019, the library of APB plug-ins has grown. The original line-up of compressors and limiters has been joined by a valve-style mixer, a hybrid EQ with analogue saturation, and a multiband compressor. The latest addition to the list is another hybrid plug-in. BOB APB is based on the Bass Optimized Bias module developed for McDSP's 6060 plug-in, which is conceptually a sort of virtual 500-series rack. The tone-shaping aspects of the APB version are still handled digitally, but it adds what McDSP's Colin McDowell describes as "headroom stuff" and saturation in the analogue domain.

Functionally speaking, BOB APB can be thought of as a low-frequency enhancer. Unlike, say, the dbx Subharmonic Synth, or a guitar octave pedal, it doesn't synthesize harmonic content below what's present in the original material. Rather, it seems to apply something more like a resonant EQ, before using compression and saturation to thicken up the resulting sound.

The core controls of both BOB APB and its native antecedent are fairly simple. Two 'styles' of bias are available, labelled simply A and B. The Freq control sets the frequency at which the process is focused, and the Bias control determines how hard it's pushed. With Bias at 0% and style A selected, the Freq control acts like a high-pass filter; as you turn up the Bias, you begin to hear a resonant 'thump' at your chosen frequency, while frequencies below this start to make it through again, albeit at a lower level. Style B is more subtle, and perhaps more like a resonant shelving band. The Squash control applies frequency-sensitive compression, with a Rate control governing release time.

Unique to the APB version of the plug-in are two saturation modes, labelled I and II. These can be used independently or together, and a single Gain control sets how hard the saturation should be driven. An output Trim control lets you rein in the signal at the back end.

To get you started, McDSP have helpfully supplied a decent selection of presets categorised by application: Bass, Drums, Guitars, Mix and More. These certainly aid the orientation process, but operation is generally quite intuitive, and it won't be long before you can do without.

Light Touch

I was pleasantly surprised to find that there are subtle and non-obvious

Low Latency Mode

Like a conventional computer audio interface, the Analog Processing Box needs to buffer the audio that it sends out to and receives back from its D-A and A-D converters. This buffering introduces a latency or delay, which is reported to the host DAW and compensated for, so not a problem at the mix. But if you want to monitor or track a live input through an APB plug-in, the total latency would be equal to that of your audio interface and the APB added together (plus some additional fixed latency on both fronts), and this can be an issue.

When I first reviewed the APB16, I mistakenly reported that its buffer size always mirrored that of the host DAW. In fact, it was fixed at 1024 samples, and hence was really too slow to ever be used on a live input that was part of a performer's cue mix. In beta at the time of writing is a new Low Latency Mode, which makes the APB operate as I thought it always did! With this installed, APB latency changes with the audio interface buffer size you choose in your DAW. At the lowest 32-sample setting available in Pro Tools, an APB plug-in now reports a delay of 207 samples, which equates to 4.7ms at 44.1kHz. On my rather modest Intel Mac Mini, this gave rise to clicks and pops, but moving to a 64-sample buffer only raised the APB delay to 231 samples, and that worked fine. Note that if you want to use more than one APB plug-in in series, the delay will be cumulative, as each one you add introduces a new round trip.

applications for BOB APB. For example, during the review period I was working on a restoration project with audio that had been recorded off the radio before passing through several generations of cassette dubbing. Some of the tracks were simultaneously woolly and overbearing in the low midrange, whilst lacking in true bass. Although BOB doesn't synthesize subharmonic content, it was able to work effectively with what little there was in the sub-100Hz region to regenerate a feeling of proper low end.

Most of the time, though, you'll want to be using this plug-in in rock, pop and electronica mixes, and here it can be devastating. I often find that synth basses are too subby, for example, and BOB APB makes it easy to rebalance the tone so that it will be audible on small speakers without robbing the synth of its impact. The saturation tends to be quite subtle unless driven hard, but can really help things cut through in the midrange.

Likewise, if you're afflicted with an anaemic bass guitar recording, or even a thin-sounding vocal track, BOB offers a surprisingly broad range of treatments. By far my favourite application for it, though, was on bass drums. There's something magical that happens when you crank up the Bias at 60 or 70 Hz before applying gobs of compression to rein it all in again. Topped off with a helping of analogue saturation, it just seems to turn almost any kick drum track into something punchy and solid that waves the rest of the mix aside contemptuously whilst simultaneously belonging perfectly in the track. It's a trick that I've sometimes been able to pull off using the Thump button on Soundtoys' Decapitator, but APB BOB makes it a lot easier to achieve.

Bass enhancement is quite a specialised process, and although I can well imagine wanting to use BOB on every kick drum track from now on, the reality is that not every mix really needs it. And in a sense, that's the beauty of the APB system. You might cavil at the idea of spending a lot of money on a hardware box that only did what BOB APB does — but as far as McDSP's Analog Processing Box goes, it's simply one more arrow in the quiver. Or should that be a feather in the cap? Either way, this is yet another well conceived and well implemented processor that proves the fundamental strength of the concept. ■■■

summary

A versatile and easy-to-use bass enhancement tool that might just be the magic bullet for weedy kick drums.

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Polyend Play+

Sample & Synth Sequencer

Polyend's Play gets a major upgrade.

SIMON SHERBOURNE

On the face of it, Play is the more approachable of Polyend's two grooveboxes, although actually neither Play nor its sibling the Tracker do things in conventional ways. The first version of Play, which we reviewed just a year ago, is a sample-powered grid sequencer, ideas generator and portable performance unit.

Play+ looks identical to Play, but has a significantly upgraded brain with the capacity for some big new features: chiefly, internal synths, stereo samples and audio over USB. Depending on how you use the Play, upgrading may not be essential, but if you are a current owner you can send back your Play and exchange it for a Play+ at half price. The original continues to be available at a reduced cost.

Re: Play

To recap, Play and Play+ offer a couple of primary ways of creating music. For programming a sequence from scratch, you use the 'pick and place' system of pre-configuring a sound, then placing it anywhere on the grid. Whether you organise yourself in tracks is up to you; each step is fully independent. Steps can be adjusted later on by selecting them, either individually or in groups. On Play, you have a page for assembling samples across eight tracks, and another for MIDI. On Play+ the MIDI page is also the workspace for adding sounds from the internal synth engines.

To really get the most out of Play, though, you need to get on board with the generative Fill way of working. This populates sequences for you, using template styles, Euclidean distribution or randomness. Multitrack drum patterns can be spun out by this process, using sounds from categorised folders. Patterns you create can be mutated by further randomness and chance actions.

It is also possible to live record melodic sequences, using a virtual keyboard to the right of the grid or a connected MIDI source. This is especially relevant now that there are onboard synths. As you can only play one sound at a time, there's no way to finger-drum a kit of drum sounds.

A Play/Play+ project has a dedicated grid page for storing patterns, but there's

also a concept of Variations, where any track within a pattern can flip between 16 versions. Multiple patterns can be selected to play sequentially. These various features can be put to use to construct and perform beats and musical ideas, and there's a Perform mode which transforms the whole grid into a palette of real-time effects including pitch transposition, step rearrangement and beat repeating, audio effects and stuttering.

Difference Engines

Play+ sports four different synth engines: ACD, FAT, VAP and WTFM. A project has three synth slots, which can be filled by any combination of these (including multiples of the same type if you want duelling 303s, for example). Sequences using the synths are more like MIDI patterns in that each step is linked to a sound source, rather than being a self-contained sound, as is the case on the sample side. So if you change the patch loaded into Synth slot 1, it will affect all steps that use that slot. 'Sound Locks', ie. recalling different patches per step, is not a thing here.

Let's look at these synth models, then. ACD is aimed at classic monosynth sounds, although it can be played polyphonically as well. It offers a mix of saw and square waves plus a sub. Although there's no per-step glide in Play's sequencer, with a bit of portamento and the RD3 filter option you can get some fairly pleasing acid lines out of it. But it's more versatile than that and can double as other Roland-y synths like the Juno 6 or SH-101.

FAT is a Moogy virtual analogue, with three oscillators. Instead of individual oscillator controls you get three 'style' controls: Fatness applies detuning, Timbre is pulse width, and Brightness seems to control the blend of oscillators. As well as the Moog-style filter, there are two Oberheim modes.

VAP moves into more programmable polysynth territory. It has two oscillators with independently controllable pitch, shape and PWM. There's an extensive selection of filters, and a lot more in the way of modulation, with three envelopes, two LFOs and a six-way mod matrix.

WTFM brings a versatile implementation of two-operator FM synthesis. The two operators are themselves scannable wavetable oscillators offering a range of timbres. They each have self-feedback routing as well as feedback from 2 to 1. Like VAP, WTFM gets a generous selection of filter and modulation options.

Programming of the synths is all done in a rather joyless scrolling list editor akin to Roland's Zen-powered devices. Most of the time you'll want to choose a starting point from the presets and make changes using the three encoders that become macro controllers when in synth mode. These can be different for each patch, but typically will align with the Filter and Envelope controls that are used for samples, with the Overdrive/Bit Depth encoder mapped to other parameters like envelope amount.

Using The Synths

Synth parts are added much like MIDI, with the three synth slots appended to the list of MIDI ports in the source selector. As well as manual step entry and recording, you can use the Fill workflow, where there are new modes for Chord, Bass or both at once. Results will depend on how many tracks you select: a single track will give you a monophonic melody, two to four tracks will result in chords, and with five tracks you'll get a bass line as well. As with MIDI, each track represents a voice, so to sequence or record polyphonically you need to select multiple tracks, which feels a bit awkward compared to sequencers with true poly tracks. However, since we reviewed the original Play the sequencer »

Polyend Play+
$799

PROS

- It has synths.
- All tracks split out to multi-channel USB audio.
- Stereo samples.

CONS

- Locked to three synth sounds per project, and all parameters locked to steps.
- MIDI and synths still excluded from Perform.

SUMMARY

Play+ offers a unique, assisted sequencer-centric workflow with greatly expanded sonic capabilities.

Class-compliant USB audio provides real-time streaming of sample tracks, synth engines and effects into your DAW.

Back-panel socketry remains the same as on the original Play, with 3.5mm audio out and MIDI I/O ports, a MicroSD slot, USB C port and the power button.

» has been updated to include a piano-roll mode which works with multitrack selections. There's also a chord mode for programming chords on a single track, and by automating the chord type per step you can quickly create grown-up-sounding progressions.

Frustratingly, Fill always defaults to using MIDI channels instead of synths unless you've specifically gone in and assigned this in the settings. Fill is also not quite as much of an ideas machine as it is on the sample side, as it will always use whichever patches you have loaded instead of a pool of sounds.

Perhaps the most significant difference between how Play+ and most grooveboxes do synths is that, like sample playback, synth parameter changes are always tied to steps. There's no easy way to do something like adjust the filter cutoff while a note is playing. If you select a track and adjust parameters, you'll hear no change until steps are triggered, and then you'll have actually written that (static) change to those steps. For this reason it probably makes no real difference that you can't control the synth parameters from external MIDI sources.

Audio Upgrade

Play+ adds support for stereo samples, and comes with a selection of enhanced sample packs to take advantage of this. For me, though, the bigger upgrade is support for audio over USB. When plugged into my Mac, the Play+ popped up as a class-compliant audio device showing a whopping 28 inputs and two output streams. My DAW supports multiple I/O devices, but if yours doesn't you could create an aggregate device, or just connect the Play+ on its own for transfer. In fact, the stereo stream to the Play+ goes straight to the monitor output and does not feed back to the inputs, so you could use it as a portable audio interface while running your Play tracks through your DAW.

The inputs include a stereo mix, all eight stereo sample tracks, three stereo feeds from the synths and separate stereo returns carrying the delay and reverb effects. This gives you complete separation for projects that you want to develop further or mix in your DAW. Really impressive! The only down sides are that it's 44.1kHz only (my studio is all clocked to 48kHz normally), and that the channels all come in pre the Perform effects, which are a not insignificant part of what Play+ can do. Likewise, audio coming from the computer also bypasses the effects.

> "While there are areas that could be improved, Play+ is a wonderful thing, very much enhanced by the powerful and great-sounding synth engines."

An update that original Play owners benefit from is stem export. You can now export the current Pattern or Pattern Chain as individual channels: eight stereo for the samples, three for synths and two for effects. This actually provides a partial workaround for the fact that the Plays don't feature resampling. You can bounce things as exports and load them back in as a sample. While we've raised the subject, although the Play+ now has audio input thanks to the USB streaming, it still cannot sample.

Final Thoughts

So, Play+ has extra hardware-powered capabilities, and both models have seen a few updates, but the general approach has stayed much the same. As I found with the Play v1, this can be pretty frustrating when coming from other drum machines or grooveboxes, as Polyend have rethought the whole approach. But if you loosen up on trying to impose your normal way of doing things, and go the Play way, fun things start to happen. In particular embracing Fill, Randomness and Chance as ways to generate and regenerate patterns frees up a bunch of time and mental load from reinventing the wheel every time you want to jam. And the Perform mode can be a more fun and fluid way to get results than manually programming lots of variations and fills.

That's not to say that all is perfect. Some gaps that Polyend had talked about addressing a year ago are still evident. MIDI and Synth tracks are not affected by all the goodness in Perform mode, except for the Stutter effect, which works on the main mix. This is particularly problematic if you wanted to use the Transpose functions. Polyend also said they were reworking the chance feature, which currently can only do one thing at a time. And it's still a faff having no USB storage mode to avoid having to remove the SD card to get samples into the unit: significant considering Play can't sample, slice or edit in the box. I also think that the inability to tweak sounds as they play takes away from Play's playability as an instrument. With no movement on some of these areas, I guess development resources went into the new device instead.

But while there are areas that could be improved, Play+ is a wonderful thing, very much enhanced by the powerful and great-sounding synth engines. I treat it like a studio companion which I can collaborate with: it helps me come up with the core beats and foundations of a track which I can then noodle over with other instruments.

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Orchestral Tools
Pēteris Vasks Strings

Sample Library

Orchestral Tools' new strings library distils the sound world of a major European composer.

Orchestral Tools
Pēteris Vasks Strings
€549

PROS
- Contains a chamber strings ensemble and five solo instruments recorded from six mic positions.
- Combines unique playing techniques invented by the composer with standard performance styles for all seasons.
- The playing is of a high standard and the church acoustic is sensational.
- You can buy the instruments separately and download their mic positions when you need them.

CONS
- Not cheap.
- May be too specialised for some orchestral sample users.

SUMMARY
Orchestral Tools' homage to Latvian composer Pēteris Vasks combines a 22-piece chamber string orchestra with five first-chair soloists. Recorded from six mic positions in a 16th Century church with stunning acoustics, the samples divide into traditional playing styles and 'gestures', special experimental and atonal techniques devised by the composer. While its subtle and intimate mood makes it unsuitable for aggressive trailer soundtracks, its unique and colourful timbres will work in many musical settings.

DAVE STEWART

Since deep-sampling every nook and cranny of the orchestra in their estimable Berlin and Metropolis Ark series, Orchestral Tools have spread their wings with a multitude of non-orchestral libraries. Notable examples include Chinese, South American and Baroque period instruments, the excellent Glory Days big band horns and Tom Holkenborg's storming brass and percussion libraries, the latter featuring the incendiary drums used in *Mad Max: Fury Road*. The company also ventured into electronic territory with their affordable Fabrik series and Alexander Hacke's experimental drones library.

Despite these excursions, Orchestral Tools have never lost sight of their core mission, and likely never will. Recent releases Tallinn and Salu (reviewed in *SOS* in August 2021 and June 2023 respectively) explore the delicate, minimal sound world of Estonian chamber strings and choirs in collaboration with Michael Pärt, son of renowned composer Arvo Pärt. The company's latest major release takes a similar tack: recorded in the neighbouring Baltic state of Latvia, it's a creative partnership with one of the country's most celebrated musical sons, composer Pēteris Vasks.

Overview

Pēteris Vasks Strings (originally 234GB in size, 96.7GB when installed) consists of small string ensembles (first and second violins, violas, cellos and basses) and their corresponding first-chair soloists. The samples were performed by members of the Grammy-winning Sinfonietta Rīga chamber orchestra and recorded in St. John's Church in Latvia's capital Riga. Like all Orchestral Tools collections released since 2019, the library runs exclusively on the proprietary Sine player, which you can use as a plug-in in your DAW or operate standalone on your desktop.

The inspiration for this project was Pēteris Vasks' acclaimed *Voices* symphony for string orchestra, which deals with the inner struggles of the Latvian people as they fought for independence from Soviet rule (see box). Orchestral Tools' goal was to capture the sound and spirit of the piece with all of its unique techniques and textures, working closely with Vasks to ensure his musical vision was correctly represented.

As well as the bread-and-butter articulations you'd expect to find in a strings library, this collection features special expressive styles invented by the composer, which Orchestral Tools call 'gestures'. While these unusual, ear-catching articulations step out of the ordinary and help give the library its unique character, the standard articulations offer plenty of scope for traditional string arrangements.

Ensembles — Sustains

In PVS (as I'll have to call it to placate my spellchecker) Orchestral Tools adopt an unusual approach with their velocity crossfades. This feature assigns dynamic control to your keyboard's mod wheel, enabling you to crossfade smoothly between instruments' dynamic layers and create natural-sounding swells, fades and crescendos. In this library the ensemble long notes' three dynamics feature varying »

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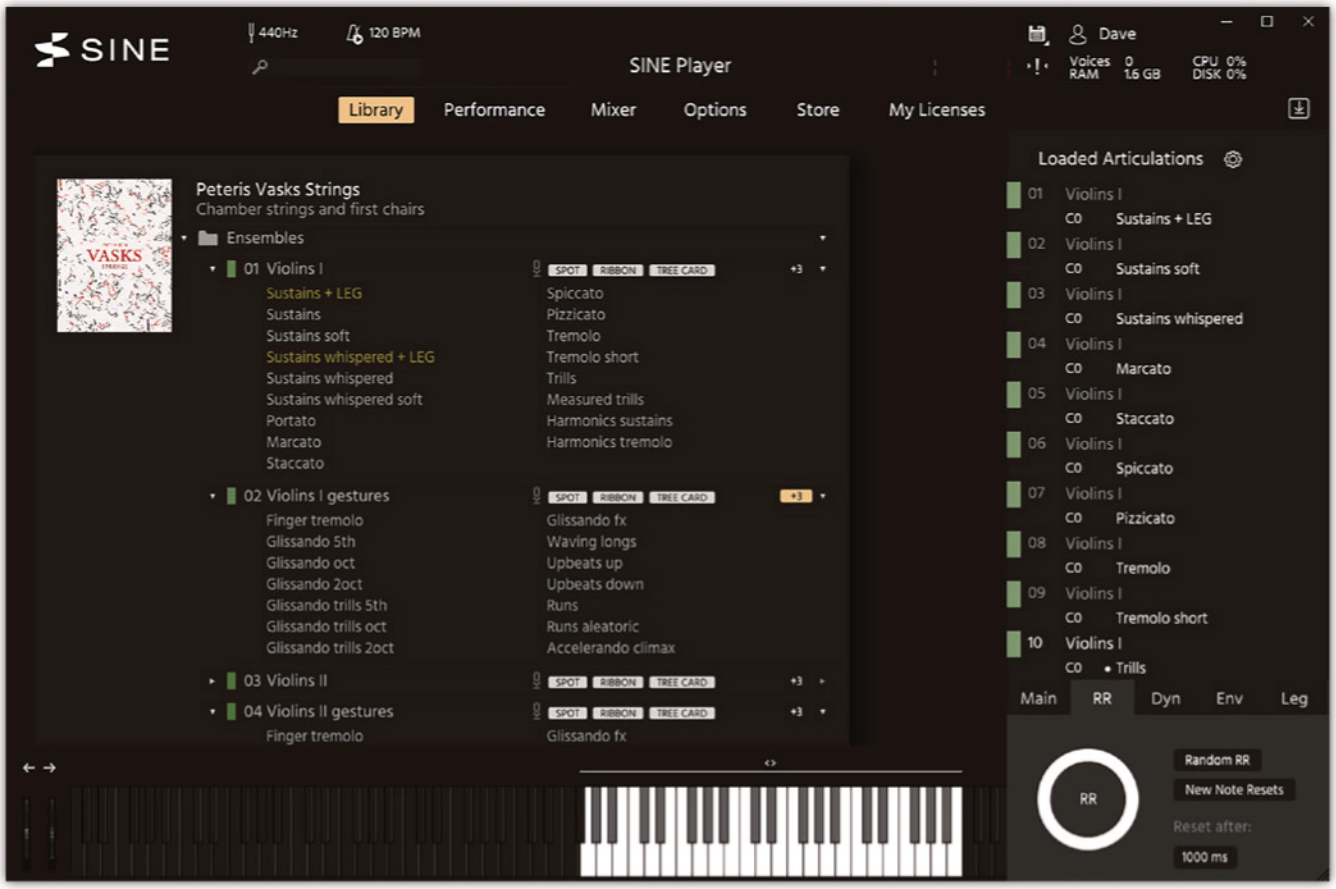

The Sine Player's interface links directly to the Orchestral Tools store, where you can download separate instruments and individual mic positions.

» degrees of con sordino muting: the lowest *pp* dynamic was played with mutes attached, half of the players removed their mutes for the middle *mf* layer, while the loudest *ff* performances were played without mutes.

Each layer also has a different vibrato intensity — the quiet notes use little or none, the middle dynamic introduces a restrained vibrato and the loud layer features a full (but not over the top) vibrato style. This means you can create expressive melody lines which start out subdued and austere-sounding before blossoming into an intense, emotive loud delivery, a transition often heard in Pēteris Vasks' string writing.

Sustains offer a choice of immediate and soft attacks along with a delicate 'whispered' sordino variant. In this, half the players play sul tasto while the other half play with the back of the bow (col legno tratto), resulting in a beautifully soft and breathy texture. The immediate sustains and whispered patches have a corresponding monophonic legato mode which does an excellent job of smoothing out melodic note transitions — I particularly liked the second violins' sweet, lyrical legato style and the graceful, dignified tone of the cello legatos.

Ensembles — Assorted Articulations

A common orchestral library fixture, the 'portato' articulation usually consists of slightly accented non-legato notes of around one second's duration. PVS' interpretation is more dramatic, introducing a pronounced volume swell which each ensemble plays at a slightly different length. The cellos' version is one of the standouts of the library, a yearning, heartfelt rendition which sounds great when played chordally.

Other deviations from the norm include trills played in minor thirds as well as the usual tone and semitone intervals, and 'measured trills' which can be sync'ed to DAW tempo. Though the latter's rhythmic pulse is fairly subtle, the metered delivery imparts order and direction to the chaotic effect of 22 string players playing trills at different speeds! I also enjoyed the tremolo short notes' fierce, attacking *ff* layer and the tremolo muted harmonics, a creepy glacial timbre at which the second violins excel.

In addition to these uncommon artics are the standard spiccato, staccato and marcato short note styles, which can be woven together into effective rhythm passages. The first violins turn in some forceful loud marcatos and their tight spiccatos are right on the money, but the violas' spiccatos lack the energy required for driving cinematic ostinatos.

While there's much to enjoy here, one performance style gets my 'best in show' award: the ensembles' pizzicatos are fabulous, collectively making a big, sumptuous sound with a huge bottom end and a crystal-clear, ultra-precise high register. Very impressive, and lots of fun to play.

Ensembles — Gestures

The 'gesture' artics are where this library shows its true colours. Many are derived directly from the scores of Pēteris Vasks, straight from the printed page to your computer via the wonders of sampling technology. An iconic example is the style described by Sinfonietta Rīga's conductor Normunds Šnē as "Tiny glissandos — like something going up to the sky, to heaven maybe." Atmospheric and haunting, this »

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unique articulation is a hallmark of Vasks' string writing.

Rather than the in-your-face sound effect heard in cartoon soundtracks, Vasks' glissandos consist of a quiet, controlled and precisely timed pitch slide. Played up and down in a choice of fifth, octave and two-octave ranges; the slide terminates with a sustained note held for two or three seconds. This style is often used by the composer to resolve phrases, and these perfectly synchronised performances also sound tremendous when played as chords.

Taking a sidestep into horror film world, the unpleasant, queasy-sounding glissando trills ratchet up the tension, while the aleatoric 'waving longs' style delivers the climax, starting as a disquieting quiet semitone trill before erupting into deranged, random up-and-down slithering when you push up the mod wheel. Meanwhile, composers working in less scary genres will enjoy the wild ascending glissando effects, a set of loud full-range slides which the players perform with gusto.

The intriguingly named 'finger tremolos' had me wondering — is this the nervous finger drumming you do on the arm of your chair while waiting to be summoned into the dental surgery? In fact, these finger tremolos are a kind of very fast trill played at two dynamics. The PVS version employs minor third, major third and fourth intervals, enabling you to create magical sustained harmonic shimmers with one key press.

The Sine Player's mixer page.

Supplementing these idiosyncratic articulations are some useful tools for adding movement to your scores. A set of upbeats (two quickly played short notes) cover all chromatic intervals up to a fifth in both directions, and work well for comedic interjections, cheeky grace notes and ear-catching accents. In a piece of musical oneupmanship reminiscent of Nigel Tufnel's "these go to 11", PVS's fast runs actually go up to 12 — that's to say, they rush up beyond the octave to a high 12th interval before quickly descending to the target octave note. The high strings do a good job performing the runs in major, minor, melodic minor and Dorian minor scales, but the cellos sound like they're struggling a bit.

The ensembles' gestures bow out with a pair of contrasting performances. The gibbering, mod wheel-driven madness of 'aleatoric runs' feels like having your head filled with wasps, but the 'accelerando climax' provides a more acceptable musical experience: it consists of a series of accelerating loud, single staccato notes which launch into an exuberant two-octave glissando up to a pre-orchestrated major chord. In the case of the basses, the glissando descends to a single low note. Startlingly original, and a suitably triumphant conclusion to this section of the library.

First Chairs

PVS's five solo instruments give users the option of a string quartet (which traditionally omits the bass) or quintet. You can also layer the soloists over the ensembles to add focus and definition. With few exceptions, the so-called 'first chairs' mirror the ensembles' performance styles and gestures so there's no need for me to revisit those artics — instead, I'll highlight the areas where the solo instruments deviate from the ensembles.

The most striking difference affects the first chairs' sustains and legatos. With these, Orchestral Tools recorded three distinct styles: lyrical (fairly quiet, moderate vibrato), expressive (loud with a strong vibrato) and sul tasto (very quiet, minimal vibrato). Rather than combining these artics in single patches with three dynamic layers, OT presented them separately. The reason given is that crossfading them would have compromised realism: understandable, given the degree of vibrato variation.

Microphone Positions

All instruments in the library were recorded from six mic positions, which can be individually downloaded. The default position is the main Decca Tree: rich and full with a tremendous bass presence, its omnidirectional mics sound great for full section mixes. By contrast, a second Decca Tree with additional cardioid mics sounds thinner and less ambient, with a tighter low end, which could be an advantage when using these strings in a pop context.

The close-up spot mics can be used to add definition to the Tree, or on their own for more intimate settings. In the first-chair solo instruments, the spots are panned across the stereo field, with the first violin on the left and cello on the right flanking the second violin and viola, while the bass is centralised behind the other players. You can reposition the instruments by adjusting their pan settings in Sine's mixer.

I've often bemoaned the Sine player's lack of a built-in reverb effect, but that's not a problem here. A set of ribbon mics and a stereo pair of so-called 'AB' mics both add a lush natural ambience to the sound, the latter providing a majestic sense of size and distance. Surround mics in the far recesses of the 13th Century church capture its unique reverb, and would be the ideal candidate for the rear channels of a surround mix.

Auditioning these mic perspectives on headphones, I was struck by St. John's Church's wonderful acoustic — it's easy to see why it's Pēteris Vasks' first-choice location for concerts and recordings of his music, and Orchestral Tools have done a great job of capturing its unique, beautifully spacious sound.

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» Each legato style has a different feel: the lyrical samples are somewhat downplayed, mournful and reflective, the expressive style is forthright and romantic and the sul tastos sound tentative and tender, their vibrato-free performances evoking the atmosphere of quietly played periodinstruments. The first violin combines a lovely timbre with a heart-melting emotive delivery, while the second violin's expressive style works well for lively folk melodies. The viola and cello soloists also contribute great heartfelt performances which benefit from their instruments' beautifully rich and vibrant low registers.

Another stylistic departure comes with the first chairs' single and short trills, in which the principal note, the note above and the principal note again are played in rapid succession. Played in a choice of minor second, major second and minor third intervals, these animated, colourful events sound great in accented stabs and chordal flourishes. Other first-chair artics which caught my ear include some fabulous glissando effects (energetic slides up the full length of each string) and the two solo violins' 'waving longs', which do a passable impression of a swarm of bluebottle flies.

"The ensembles' pizzicatos are fabulous, collectively making a big, sumptuous sound with a huge bottom end and a crystal clear, ultra-precise high register. Very impressive, and lots of fun to play."

Conclusion

This library hits its targets and maintains high technical and musical standards throughout, with instruments playing across their full range wherever practical. Though there's very little to criticise, I missed the provision of 'full strings' patches that lash the ensembles together into single playable patches, as these can be a useful tool for sketching and composing. That said, I found that simply layering the five ensembles creates a convincing full symphonic strings sound of surprising size and grandeur, which, in conjunction with the lush natural ambience, wouldn't sound out of place in a pop arrangement.

It would be wrong to think of this as a purely experimental library: though some of its unusual techniques and atonal excursions suggest an *avant-garde* mindset, they function in Pēteris Vasks' works as expressive devices which add to the music's emotional impact. Although I wouldn't recommend it as a default strings library for aggressive trailer soundtracks, its mix of traditional and non-standard playing styles should appeal to imaginative composers of all genres with the confidence to step into a new sound world and add something genuinely different to their scores.

$ Full library €549. Individual ensembles: First violins, Second violins, Violas, Cellos €92 each, Basses €69.
$ Individual First Chairs: First violin, Second violin, Viola, Cello €69 each, Bass €54.
W www.orchestraltools.com

Pēteris Vasks

"I am a happy person", chortles Pēteris Vasks. "I live in the most beautiful country in the world, Latvia, and to top it off, I am a composer, and I don't know anything more beautiful in this world than music, and that's why I am a happy person in my beloved homeland."

It's fair to say this amiable, down-to-earth and nature-loving man has experienced less happy times. Much of his life was spent with his country under Soviet rule, enduring years of severe political, religious and cultural repression. Vasks says of this period, "It was a time when life under this occupying force had taken us so far that we were close to simply disappearing as a nation." After almost a half-century of struggle, Latvia finally declared independence from the Soviet Union in 1990, inspiring Vasks to compose one of his best-known works: the celebrated *Voices* symphony for string orchestra, which in turn inspired Orchestral Tools to create a sample library reflecting the piece's sound, styles and spirit.

Born in 1946, Vasks trained as a violinist and double bass player and worked as an orchestral musician before entering the State Conservatory in Vilnius, Lithuania to study composition. Though his early style was influenced by the aleatoric experiments of Witold Lutosławski and Krzysztof Penderecki, his music is rooted in tonality — dissonance is carefully managed, and stands in contrast to a strong, pervasive sense of harmonic resolution.

Other important works include *Cantabile* (1979) and *Musica Dolorosa* (1983) for string orchestra, *Distant Light* (concerto for violin and strings, 1997) and the beautiful *Concerto For Cor Anglais & Orchestra* (1989). Three times winner of Latvia's prestigious Grand Music Award, Vasks was also awarded the MIDEM Cannes Classical Music Award for *Distant Light* and his *Symphony No 2* in 2005. More recent compositions such as *Bass Trip For Solo Double Bass* (2003) and *Sonata Estiva For Solo Violin* (2022) see a return to the composer's first instruments. His music has also been used in several film and TV soundtracks, including Dzintra Geka's *Birds Of God* and Gunars Cilinskis' *The Heiress Of Werewolves*.

A happy life lived through difficult times, eventually emerging into a digital age which would have seemed unthinkable in the dark days of Soviet repression. While some classical composers shut themselves off from today's music technology, Pēteris Vasks sets a positive example by sharing his musical ideas in sampled form with the wider world, thereby expanding and enriching his immense musical legacy.

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Softube Console 1 MkIII

Software Mix Environment & Hardware Controller

The latest version of Softube's Console 1 gives you even more reasons to abandon the mouse.

SAM INGLIS

Console 1 is now 10 years old. You might expect such an enduring success to have inspired copycat launches from other manufacturers, but not so: Softube's first foray into hardware has remained almost unique. That, I think, reflects the one-off blend of expertise, imagination and determination needed to create something that cuts across so many product categories.

You can read our June 2014 review of the original Console 1 at www.soundonsound.com/reviews/softube-console-1, and it's probably worth doing so if you're new to the concept. To provide a brief recap, the idea is to replace DAW mixer functionality with Softube's own, based around the emulation of classic large-format console processing. This is implemented using software and hardware components. First of all, there's a Console 1 plug-in, which you instantiate in the last insert slot in every channel on your DAW mixer. Second, there's a control surface with knobs and buttons that map precisely onto the plug-in parameters. Tying it all together is an application called Console 1 On Screen Display, which runs in the background on your computer.

When it first launched, Console 1 was almost completely independent of the host DAW. Softube have since worked with software manufacturers to achieve deeper integration. Almost all DAWs now support the ability to transfer track names and numbers into the Console 1 plug-in automatically, and many go further, for instance by allowing the controller's mute, solo, volume and pan controls to manipulate those functions in the DAW mixer.

Console 1 users today also benefit from a decade's worth of plug-in development. At launch, the system was based around a licensed emulation of the SSL E-series console. Over the years, numerous alternatives have been added, and it's possible to mix and match on a per-channel basis if, for example, you prefer API-style equalisation with Neve-style compression on vocals. A tie-up with Universal Audio means Apollo and Satellite owners can use many of their UAD Powered Plug-ins within Console 1, too, although you may not get access to all parameters.

At around the same time UAD support was added, back in 2017, a MkII version of the hardware was released. This was largely identical to the MkI, but a move to Chinese manufacturing made it much more affordable. Now, the Year of the Dragon brings us the biggest update yet.

On Your Marks

The hardware controller is now known as Console 1 Channel, to distinguish it from the companion Console 1 Fader surface. It has enjoyed a ground-up redesign, and although it's about the same width as its predecessor, it is a fair bit deeper. The layout is similar, with a row of 20 track selection buttons along the top and the EQ controls in the centre, but there are extra controls and, most strikingly, a pair of high-resolution colour displays. The cosmetics and physical design are all-new and very classy, with a robust aluminium shell, expensive-feeling rotary encoders, super-bright white LED rings and understated panel legending. It is all very monochrome, though; a bit more colour might have made the different sections more easily distinguishable at a glance. The only real ergonomic gripe I can muster is that the buttons used to exit menus are awkwardly positioned above the encoder used to scroll through them.

The brighter LEDs and displays mean Console 1 Channel MkIII requires more current than its bus-powered predecessors. For this reason, it has two USB-C ports discretely hidden either

side of the centre section. In theory, it too can be bus-powered, but in practice, I got low power warnings both driving it directly from my Mac Mini and from the 'high power' port on a powered USB hub. Softube have evidently anticipated this, because they supply a PSU with a nice long USB cable that can be connected to the second port. Software installation and firmware updates are handled, very smoothly, by Softube's own Softube Central utility.

Other Views

One of the basic ideas behind Console 1 is that, in normal use, you don't look at the plug-in window itself. Instead, the On Screen Display running in the background detects any control input and automatically flashes up the Console 1 interface as an overlay on whatever else you're looking at. Leave the controls alone for a few seconds, or hit the Display button, and this overlay disappears again (this behaviour can, of course, be modified). As long as you also have a transport controller for your DAW, this means you rarely need to touch the mouse and keyboard. Conversely, it also means Console 1 stays out of the way when you are doing something that's outside of its remit, such as audio editing or selecting a reverb plug-in.

This intended way of working meant there was no need for displays. If you wanted to know which track was currently under control, all you'd need to do is move a knob or press a button, and the overlay would appear on your computer screen. Nor did it matter whether you were in your DAW's mix or arrange page, or indeed a completely different application. That original way of working has been enhanced even further in MkIII, because the new knobs are touch-sensitive: simply resting a finger on an encoder is enough to bring up the overlay without making any parameter changes.

So why add screens in MkIII? Well, Softube say that most of the changes they've made are in response to user feedback, and Console 1ers have found other ways of working with it. It's possible, for example, to have the overlay remain invisible unless you hit the Display button, or to have it displayed in a minimal Scribble Strip view. So clearly there are people out there who want to keep the visual focus on their DAW even when adjusting Console 1 controls, and the new displays certainly make that easier. The left-hand display can be set to show either track name and input signal level, or an EQ graph with animated spectrum analyser display. Turn an EQ knob and a readout of the parameter name and value appear. The right-hand display offers either track name and output signal level, or a visualisation of the compressor response with gain-reduction metering and similar parameter readout. All of these displays pretty much mirror the way the same information is presented in the overlay, which is pleasingly logical. They're sharp, bright and easy to read, albeit on the small side.

Doubling Up

User feedback has also led Softube to extend the functionality of the Console 1 plug-in and, by extension, the controller. The original was designed to implement four EQ bands and a single compressor per instance, along with separate high- and low-pass filters, a Shape section encompassing an expander/gate and a transient shaper, and a surprisingly versatile Drive section. That sounds like a powerful toolset, and it is, but some users found that having only a single EQ and compressor stage could be limiting. In particular, you might have good sonic reasons for loading one of the optional vintage emulations into these processing stages, yet simultaneously wanting the detailed control that is available in a more modern design. You might, for instance, wish to perform general tone shaping using a Pultec-style equaliser, whilst also notching out a resonance in a way that isn't possible on a Pultec.

So, in MkIII, the five processing stages of Console 1 have become nine. One of the new additions is tucked away at the lower left. A button labelled Tape/Preamp introduces, you guessed it, tape or preamp modelling, and an Input Gain knob determines how hard this should be driven. This isn't level-compensated, so turning up the input gain increases the level feeding the other stages, »

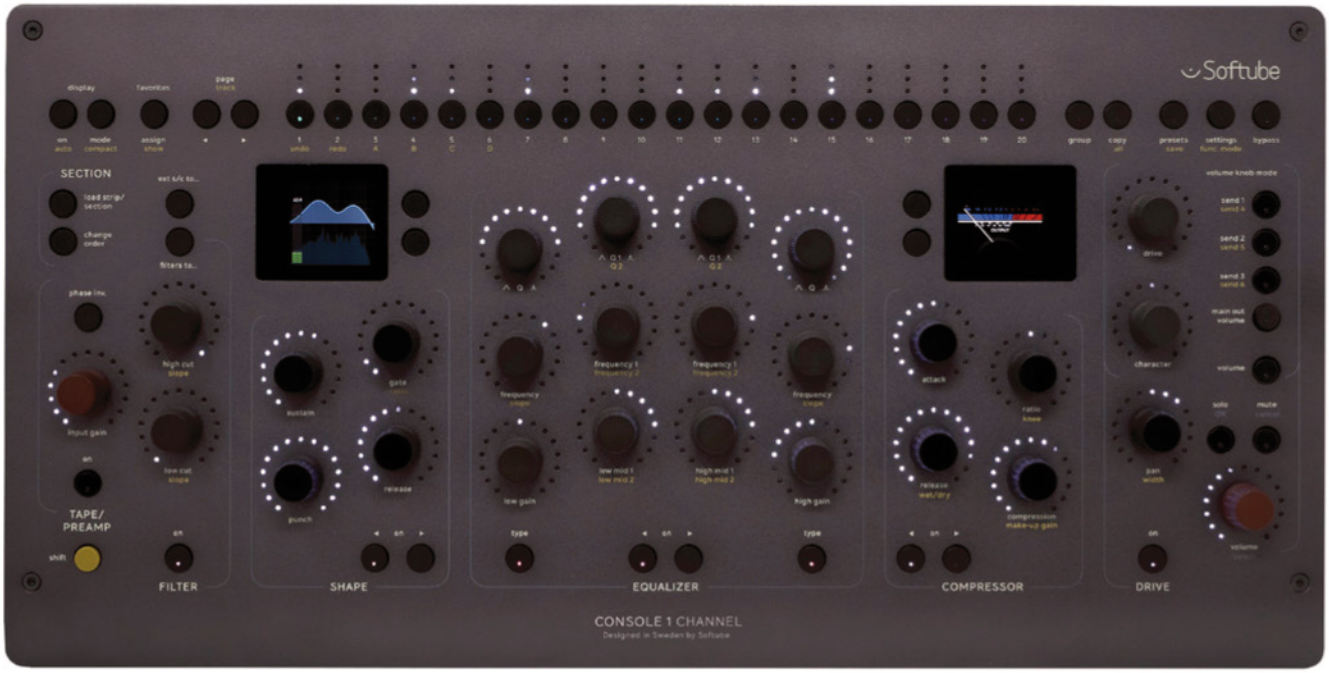
Softube Console 1 MkIII
$899

PROS
- The new Console 1 Channel controller is a beautiful object, with high-quality screens and encoders.
- MkIII adds new Tape/Preamp section and doubles up the Shape, EQ and compressor stages.
- There's now a wide selection of emulated processing options, including many UAD plug-ins.
- Lots of smaller improvements.

CONS
- Big price increase compared with MkII.
- Much less satisfying in Pro Tools and Logic than other DAWs.
- Some of the newer functionality isn't quite as streamlined as the core feature set.

SUMMARY
The third incarnation of Softube's Console 1 goes upmarket, pairing a highly desirable hands-on controller with a beefed-up software mixer.

The main On Screen Display overlay mirrors the layout of the Console 1 Channel controller, except that the track listing is moved to the bottom.



The Console 1 MkIII is a little deeper than its predecessors, measuring 435 x 45 x 219mm.

» even with Tape/Preamp switched off. The default tape emulation offers the same A, B and C algorithms available in Softube's standalone Tape plug-in, and makes a nice subtle counterpart to the more obvious Drive stage in Console 1. Tape/preamp modelling always takes place on the input side, whereas Drive is on the output side.

That makes six stages, and the total of nine has been reached by doubling up the Shape, EQ and compressor. In other words, you can now simultaneously load two different gate/transient processors, EQ and compressor types — so if, for example, you want to pair your Pultec with an SSL, or use the classic 1176/LA-2A combo on a vocal, you can do so entirely within a single Console 1 instance (I'm not sure what the use case for loading two transient processors is). New buttons toggle between the two, with integrated LEDs indicating whether or not they're active. Since this development has been prompted by user input, there's obviously a demand for it, and it's been implemented very smoothly and without breaking backwards compatibility with older Console 1 Channel controllers. It's also supported by a pretty generous selection of vintage and modern offerings in the bundled Core Mixing Suite, with lots more available as cost options. Speaking for myself, though, I've always felt that Console 1 can be a tempting pathway into the sin of over-processing things at mixdown, and MkIII makes that pathway quite a bit broader!

There are now numerous options that can be loaded into the two EQ and compressor stages.

Shifting Responsibilities

There are lots of other, more minor improvements. On MkI and MkII, the Shift button engaged a Fine Adjust mode for the encoders. This is no longer required with the new high-quality encoders, which has allowed Softube to give many of the knobs shifted functions. On the original, for example, compressor make-up gain was automatic and couldn't be defeated. Holding the Shift button and turning the Compression dial now allows you to set make-up gain manually. The Ratio control has likewise gained a shifted alternative function as a Knee control, while the filter frequency controls can modify filter Slope in conjunction with the Shift button. The outer EQ bands now boast additional Q controls that allow them to be fully parametric like the two inner bands, further increasing the processing power on offer.

Things have also been rearranged on the right-hand side, so instead of the Drive, Character and Pan knobs doing double duty as send controls, there are now buttons that reassign the main Volume control to a specific send. At the top, meanwhile, the 20 channel select buttons are augmented by a miniature meterbridge composed of three LEDs per channel, but my favourite new feature is that these buttons now have multi-coloured LEDs integrated into them. In compatible DAWs, this allows the colours of the Console 1 Channel track selection buttons to mirror those of the mixer channels in your software — a surprisingly huge boost to navigation.

Also new, at least since I last looked

at Console 1, is a system of Favourites. Even with 20 track selection buttons, navigating large projects with a small controller can require a lot of banking up and down. Favourites lets you pick your selection of the most important tracks and assign them to a separate, smaller track list. You might, for example, make Favourites of the group channels in your project, or lead instruments such as vocals and solo guitar. Switch to Favourites mode and the track selectors show only the channels you've made Favourites. It's both simpler and, to my mind, more useful than the Layers system introduced with Console 1 Fader.

The Console 1 user experience has always been somewhat DAW-dependent, and that remains the case. Integration with PreSonus Studio One, for example, is absolutely superb. Apparently small things like having the track selection buttons pick up track colours from the Studio One mixer make a huge difference in practice, as does the ability to control Studio One's aux sends. In Pro Tools, by contrast, Console 1 can't control sends, pan or fader levels in the DAW mixer, and attempting to solo anything can be frustrating. The difference is big enough that I think it's worth buying and learning a DAW that has good integration in order to get the best from Console 1. In practice, this seems to mean most DAWs that support the VST3 plug-in standard, leaving Pro Tools and Logic as the poor relations.

Growth Strategy

As launched, Console 1 had a purposely limited feature set. It didn't do everything — but what it did do, it did faster and more ergonomically than a mouse.

In the ensuing 10 years, lots of new features have been added, partnerships have been forged, and a companion fader surface has been launched. Now, MkIII makes the system vastly more powerful. The new Channel controller is also a big step up in terms of physical design, thanks to improvements like the screens and the perfectly weighted encoders. However, it's a step that comes at a jarring financial cost, more then reversing the price cut that came with MkII. Has all this development also come at a price in terms of complicating the original Console 1 vision?

I think that is an argument that you could make in relation to some features that weren't present in 2014. Ergonomics always take a slight hit when shifted functions are introduced, while loading a different EQ or compressor model or changing the order of processors involves pressing a couple of buttons, scrolling through a menu using the Volume dial, then pressing the awkwardly placed OK button to confirm (plus Shift again, if you want the new EQ or compressor to inherit any settings you'd made using the previous one). It's not difficult, but it's no longer faster than using a mouse.

But the great thing is that there's nothing stopping you from using MkIII in exactly the same way as MkI, accessing only the default processing and ignoring the duplicate Shape, EQ and compressor stages. If you stick to the core features, there's no shifting or menu-diving involved, and the sense of immediacy that is so central to the Console 1 experience is preserved. You'd have to ask yourself whether the slick new hardware, software developments and improved DAW integration still justify the price hike if you aren't going to exploit the extended features in MkIII, but they all contribute to making the core experience even better than it was. I look forward to seeing what Softube's imagination brings to the next decade of Console 1 development.

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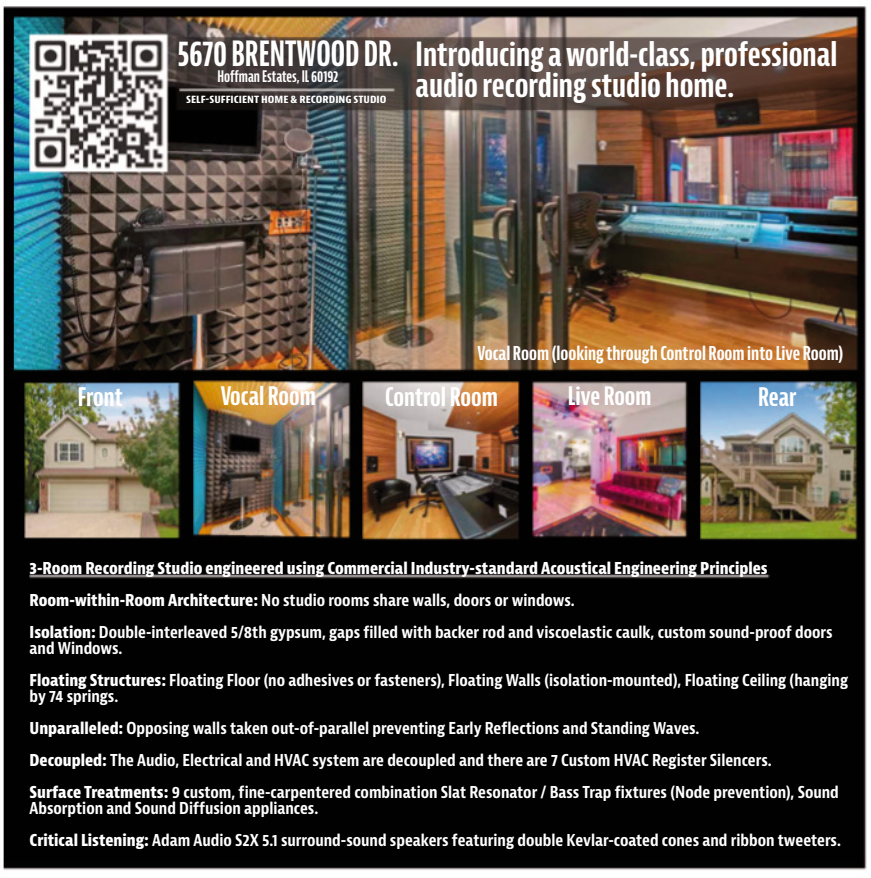
ALTERNATIVES

I said at the start of this review that Console 1 remains almost unique. I said "almost" because it does now have a direct rival. **Solid State Logic**'s **SSL 360** ecosystem includes something called the **SSL Plugin Mixer**, which runs in the background in much the same way as OSD, and a controller called the UC1 which maps very closely onto the parameters in various SSL channel strip plug-ins. At present there are only two of the latter, but by the time you read this, a new 360 Link option should allow third-party plug-ins to be assimilated into the system. Of course, there are differences: the UC1 has a dedicated bus compressor section, but lacks the Shape, Drive and Tape/Preamp stages from Console 1, as well as the MkIII's ability to duplicate sections. I haven't yet had the opportunity to try the UC1, but hope to do so soon.



Meris MercuryX

Stereo Effects Pedal

Although, at heart, this is 'just' a stereo reverb pedal, it's capable of some very unusual results...

PAUL WHITE

The world is awash with reverb pedals, but if you want to go beyond the boundaries of conventional reverb and head out into sound-design territory, I'm pretty sure that the MercuryX from Meris will tick all your boxes and then some. Meris tell us that the MercuryX is built upon their Mercury7 (reviewed by Bob Thomas in *SOS* August 2018: https://sosm.ag/meris-mercury-7), but that it includes both some additional algorithms and some very imaginative processing and routing options. While all the studio classic reverbs are here and modelled to a very high standard, it's the more creative reverb algorithms and processing combinations that have really had me struggling to confine my bank card to my wallet. And since the operating levels can be switched so as to accommodate guitars and synths or line-level sources, it could appeal to guitarists, keyboardists and mix engineers alike.

In addition to the choice of eight reverb algorithms (more on those later), there's a stereo pre-delay of up to 2.54 seconds with the option to add feedback. Each channel of the pre-delay can be set independently to different note values if required. Why such a long pre-delay? The delay lines can be modulated, HF damped, filtered and configured for cross-feedback, with routing that allows any of the processing elements to be inserted into the pre-delay's feedback loop as well as before or after the pre-delay. This routing flexibility enables the core Reverb Structures, Types and Processing Elements to be combined in numerous ways to create uniquely creative effects, and is key to the creation of some of the less obvious reverb treatments. Meris have also created what they call a Deep Modifier Section that enables control signals such as LFOs or envelopes to be routed to specific processing parameters, rather like a synth's modulation matrix. There's a Stereo Freeze function and a Hold Modifier Switch for ringing the changes during performance. Unusually for a reverb pedal, you also get a tuner.

Hardware

The MercuryX is a fairly large pedal, with four footswitches, six generously spaced control knobs, and a colour display in the centre that provides both graphical and text information. The inputs and outputs, located on the rear panel, are dual quarter-inch jacks, and make possible any permutation of mono or stereo in and out. While the pedal's

Meris MercuryX
$599

PROS
- Excellent sound quality.
- Capable of some truly dramatic treatments that go well beyond conventional reverb.
- True stereo operation.

CONS
- None (though software editor not yet available at the time of writing).

SUMMARY
The MercuryX is a reverb pedal for those wanting to go far beyond the emulation of natural spaces — although it does those very well too.

effects are digital, a great deal of attention has been paid to the sound quality of the analogue stages, which include an Analog Devices JFET input stage and an analogue mix bus. On the digital side of things, this pedal has 24-bit A-D/D-A conversion, and the 32-bit floating-point DSP runs on an ARM processor. Power requirements are 9V, centre negative at 300mA or more; there is no included PSU, but most people using this class of pedal will probably have a good pedalboard power supply.

The hardware can store 99 presets, all of which can be edited, and there's a software control/librarian app on the way too (it wasn't available at the time of writing, but I expect it to be out by the time you read this). The presets are saved within the pedal as 33 banks of three presets each, and all are directly accessible using the footswitches, which are used in pairs for changing banks and then for selecting the required preset within a bank. There's also a very handy favourites preset bank, to give you fast access to three 'go-to' presets.

In addition to the audio I/O, there's MIDI In and Out on standard five-pin DIN sockets, plus a USB port for firmware updates. The USB port will also support connection to a Mac or Windows machine running the planned Meris prEDITOR, as Meris's desktop editor/librarian package will be known. This was still in beta testing as I wrote this review, but Meris expected it to be available in the first quarter of 2024. MIDI Continuous Controller parameter control is supported, in addition to patch selection and MIDI Preset Send and Receive. MIDI Clock is recognised and is used when sync'ing relevant effect parameters to note values. There's also an Exp jack socket that allows a connected expression pedal to adjust up to six parameters at the same time.

Orientation

Despite the obvious complexity of what is going on under the hood, the user interface is very streamlined, with dedicated controls for Decay, Pre-delay, Mod, and Mix, plus three additional rotary encoder known as C1, C2 and C3, which are described as the main navigation controllers. By default, the Preset page is shown when the pedal is powered up. This is the Graphic View, in which the circle on screen shows both the preset's name and its number. Two favourite parameters for the current preset are adjusted using the two controller knobs below the display, and once it's been changed, the preset can be resaved using the 'save as' function. Graphic View focuses on one block and/or parameter at a time, but the alternative Text View allows six parameters to be adjusted at the same time, using six of the knobs. Switching between Graphic View and Text View is done in the Global settings.

Pressing the Preset/Pages controller gets you into editor mode, for which the screen shows several circles with names of sections that can be edited. The desired section can be selected using the leftmost encoder, while the lower right controller can be used to adjust values and the Preset/Pages controller cycles thought the available processing blocks. The Preset/Pages controller can also access a different edit level showing global and system settings and information. While this is very neatly thought out, I'm looking forward to the editing software becoming available, as »

The MercuryX supports both MIDI and expression pedal control, while the USB port will also cater for a Mac/Windows editor/librarian app.

» I suspect that could make patch creation a lot easier, especially given the manifold possible routing options.

When adjusting Decay, Pre-delay, Mod or Mix, the so-called Tactile Pop-Up view is shown for a few seconds, and reveals the precise values as four bar columns. This can be helpful, though for those who find this sort of dynamic interface behaviour a distraction, it can be disabled or made persistent in the global settings. Favourite parameters can also be assigned to C1 or C2 in the Edit Page.

When it comes to setting up Modifiers, each of which can be assigned to a specific parameter, the user can set how quickly changes occur, the range over which they occur and also the magnitude of the changes. The LFO A, LFO B, sample & hold and Sequencer all have speed parameters. The Envelope Modifier has Attack and Decay Time adjustments. Note-division sync of modifiers is also possible, and when this is set it overrides the modifier's speed parameter. The Hold Modifier is an envelope generator that can be triggered using the rightmost footswitch, and may be set to latching or momentary operation. Alternatively, this switch may be used to set tap tempo rates.

Reverb Algorithms

The MercuryX includes Ultraplate and Cathedra algorithms derived from the Mercury7, which, designer Angelo Mazzocco tells us, were inspired by the moody soundtrack to the film *Blade Runner*. The 78 Room, Plate and Hall algorithms are informed by classic studio hardware, while Spring, Prism and Gravity are new for MercuryX.

Ultraplate is a plate emulation, and the user gets control over all key parameters plus modulation. Cathedra is an ethereal algorithm, with a slow buildup. Spring recreates the classic spring reverbs found in some guitar amps, while 78 Room emulates the ambience of a real space. 78 Plate has a rapid build-up and a dense decay, and 78 Hall emulates a large hall with a medium build-up of reflections. Prism synthesizes geometric rooms based on settings for size, distance and reflectivity, and includes a configurable modulation section that's independent for the two channels. Gravity? Well, it's a distinctive sound, but I'll have to go with the handbook description for this one: "The Gravity Structure works by accelerating individual windows of your signal and stretching them across the horizon." Poetic, or what? Its modulation section includes pitch manipulation.

Most of the special reverbs offered by MercuryX are created by using processing in and around the pre-delays, before the sound hits the actual reverb algorithm, but there's plenty of flexibility and you can deploy these processors pretty much anywhere in the signal path, including after the reverb blocks. As the pre-delays can be pretty large, this allows for the creation of some long reverbs that change character in interesting and sometimes unexpected ways as they develop. There are lots of options, including a generous dynamics section that includes compression, swell (slow attack), a limiter, diffusion with adjustable density and Freeze, which sustains the reverb tail indefinitely. There's preamp modelling to emulate various types of analogue circuitry, as well as resonant filters and pitch-shifting in both a smooth polyphonic shift mode and a lo-fi mode. There are numerous modulation effects too, such as chorus, vibrato, vowel filtering, tremolo and something called Hazy, which adds some lo-fi pitch wobble.

"While the MercuryX nails the classic studio reverb sounds perfectly, other premium pedals can manage that too, so for me it's the special-effect reverbs that really set this pedal apart."

Special Effects?

Even in mono, this pedal sounds superb, but heard in stereo it is even more impressive! While the MercuryX nails the classic studio reverb sounds perfectly, other premium pedals can manage that too, so for me it's the special-effect reverbs that really set this pedal apart. In fact, I can honestly say that I've never heard special reverb effects as inspiring as these from any other pedal, regardless of cost.

The kind of thing you can expect is a reverb that starts off fairly normally but then fades in a sub-octave reverb a second or two later and then, after another second or two, evolves into an upper-octave shimmer reverb. Or maybe the reverb tail has a lo-fi wobble to it, or even plays a pitched sequence. Then there's the Hold/Modifier button, which can be used to freeze reverb tails or to bring about other changes during performance, such as introducing pitching effects. Bring an expression pedal into play, and the MercuryX become an extension of your instrument, offering lots of performance options.

It's the way the different reverb flavours and treatments morph into each other in such a natural-sounding way that makes the special-effect reverbs so pleasing. The deep MIDI control is also to be applauded, and once the editor software becomes available I'm sure that will make patch creation a lot more intuitive. But even if you are not into deep editing, the presets cover a lot of ground already — you may well be able to get the effect you desire just by tweaking the presets, rather than building patches from scratch.

In short, there can be no doubt that the MercuryX is a remarkable device, and one that lends itself perfectly to sound design, cinematic soundtrack creation and of course ambient music styles.

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iZotope Nectar 4 Advanced

Vocal Production Plug-in Suite

If your vocals require sweetening, a little Nectar might be just the thing....

JOHN WALDEN

The Vocal Assistant view offers an easy point of entry to Nectar 4's powerful processing possibilities.

iZotope Zotope are perhaps best known for their powerful Ozone mastering and RX audio restoration suites, but their product catalogue is actually very extensive. One of the other standouts is Nectar. Like Ozone and RX, this comprises a 'mothership' that's populated with a number of powerful processing modules, but in this case, everything is targeted specifically at vocal production. And just as they have with Ozone and RX, iZotope have been incrementally adding AI-based assistance to make all the power within this suite much more accessible to the less experienced (or more time-pressured) user.

Nectar 4 is the first update since iZotope joined the Native Instruments family, and it's available either as an individual purchase or within Native Instruments bundles such as their Music Production Suite 6. All the previous core Nectar functionality remains intact, but plenty of features have been improved or reorganised, and there are a number of new options and AI enhancements. I'll focus here on the newer elements, and on the flagship Nectar 4 Advanced version of the software, but it's worth noting that there are more affordable cut-down Elements and Standard editions too. I will try to indicate which features are available in which versions.

iZotope
Nectar 4 Advanced
$299

PROS
- A comprehensive and powerful vocal production tool.
- AI assistance gets you results very quickly.
- 'Lite' versions available for less.

CONS
- None.

SUMMARY
In its various editions, Nectar 4 can offer something useful for almost every level of experience. It's an efficient and powerful tool for getting your vocals to a mix-ready starting point with a minimum of fuss.

Meet My Assistant

For those happy to take a little AI input to get them started, Nectar 4's Vocal Assistant has had a very useful revamp and is now similar in style to that found in Ozone. As such, Nectar's main mothership plug-in window now allows you to tab between Assistant View and Detailed View. When you first initiate Nectar 4 on a track, it requests some audio input so that it can 'learn' the sound of your vocal. Then, after a brief pause while it analyses the vocal characteristics, the Vocal Assistant tab, which has a helpfully compact control set, is displayed. In the background, Nectar has actually built a complete signal chain based around its component modules, and you can see all of that if you switch to Detailed View. But this new Vocal Assistant view makes it super easy to make adjustments to the key parameters in that chain.

In the centre of the display, the frequency spectrum of your vocal is shown in white, underlain by a target reference envelope in yellow. Some standard vocal references are available but, as described in more detail below, you can also capture a reference from another track if you have a specific tonal target in mind. The Shape control adjusts the extent to which Nectar's EQ module attempts to match the tone of your vocal with that of the selected target. Intensity, meanwhile, is linked to the threshold in the compressor module. At bottom left, you can apply reverb, delay and 'dimension' (modulation-based effects). The lower right quarter is where you can experiment with various presets for the Voices module, which creates artificial harmonies based upon your vocal and pitch correction, or the Backer module, which generates alternate virtual voices to double your vocal.

I'll say more about these latter two below but, if you don't want to delve deeper, or just need to tweak Nectar's suggestions slightly while you continue developing the overall track, Vocal Assistant provides a very easy and effective point of entry.

Through A Lens

All editions of Nectar ship with a separate application called Audiolens. This will audition any audio source and, under the hood, isolate the vocal stem before analysing its tonal properties, creating a Reference file for that vocal and adding it as a Vocal Assistant user preset. Audiolens will happily listen to audio that you're streaming to your computer (you will be walked through the installation of

the necessary drivers for this on first use) so, if you want to create tonal references from a specific track by your favourite artists that's easily achieved, whatever form of audio you have access to.

In use, the Audiolens/Vocal Assistant combination is interesting and can be educational too. Of course, trying to force a tonal match between (for example) a male baritone and a female soprano might not result in the most natural result, but if you're just trying to nudge your vocals gently towards the tonal character of a specific reference, Nectar 4 can certainly give you a decent starting point — and generate a suitable signal chain — to push against.

On The Level

When Advanced edition users do switch to Detailed View, they'll find that the new Auto Level Module (ALM) is now available at the start of the signal chain, and this provides an alternative to compression as a way of ensuring a consistent vocal level. While it's not used in quite the same way, it seems similar in concept to Waves' Vocal Rider.

With ALM, the Learn process allows the module to assess the dynamic range of the source signal, and it then automatically configures itself to produce a more even vocal level. You can control the Target level and the Mix, while an extra tab that can be popped open presents further options, including

Detailed View provides a full range of configurable vocal processing modules.

In the Advanced edition, the new Auto-Level Module provides an alternative to compression to achieve a consistent vocal level through your whole track.

the clever Tame Noise, which ensures non-sung vocal noises aren't exaggerated by the auto-levelling process. You can also set a maximum range of operation for the levelling and the speed with which the levelling is applied.

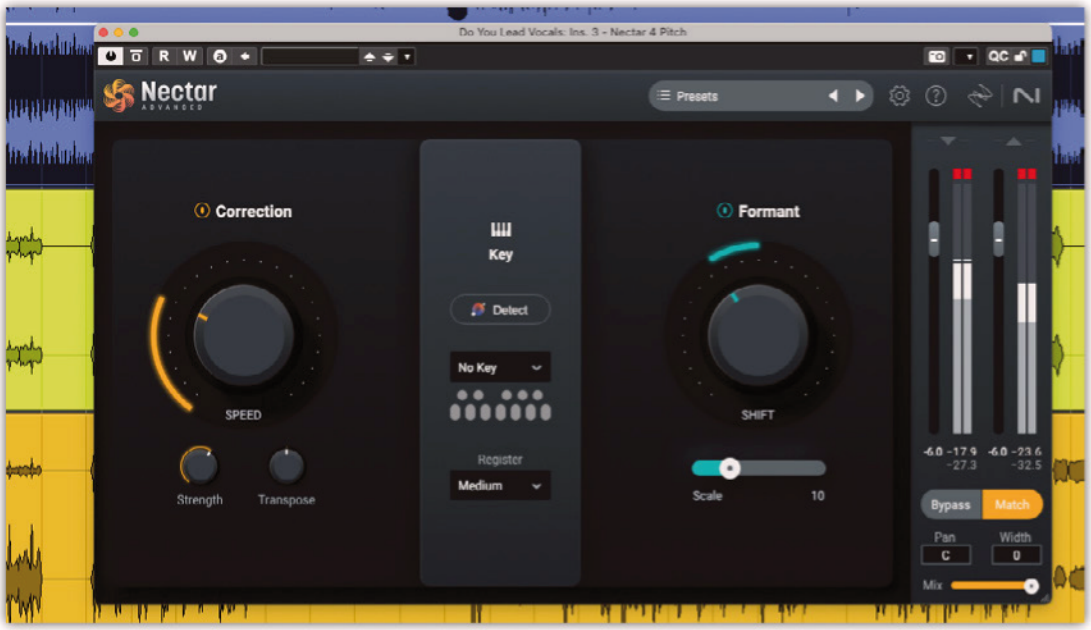
Finally, if your overall project has fairly wide dynamics, you can use the side-chain input to feed a guide signal into the ALM (for example, from a global instrumental group bus) and the module will use this to make your vocal level follow the overall dynamics of the project. Oh, and while we're on the subject of making sure your vocal doesn't get lost in a mix, Nectar's existing Vocal Unmask feature, which used to be part of the Vocal Assistant, has now been treated to its own tab, accessed from the topmost control strip.

Sweet Harmony?

In previous releases, Nectar's Harmony module allowed you to generate harmony vocal parts from your main lead vocal. That functionality has been enhanced in this release, and it now resides in the new Voices module. Perhaps more interesting is that there's a new additional module called Backer, which will generate versions of your lead vocal as if it had been sung by different singers. Happily, both of these modules are available in Elements, Standard and Advanced editions, meaning that all Nectar users have access to this feature.

Creating your artificial harmonies using the Voices module provides a microcosm of Nectar's overall workflow: it offers assistance to those who need it, but it allows detailed manual control for those who wish to configure things more precisely for themselves. For example, for Voices to do its thing, you first need to confirm the key of your track. You can select this manually if it's known, but the Detect button will let Nectar listen and make a suitable suggestion. With the key locked in, you can then simply move through the harmony presets and adjust whether you want those harmonies above or below your lead voice. These presets provide fixed harmonic intervals (unison, octaves, thirds, and so on) and more dynamic, style-based presets (moody, uplifting, lush and the like).

The panel on the left provides simple options to customise the tone, pitch correction and pitch/timing variations applied to the harmonies, while the Solo switch at top right lets you hear just the harmony parts as you tweak. It is, frankly, incredibly easy to create harmony parts that, sat at a suitable level in your overall mix, will be genuinely convincing. »



Advanced users get many of Nectar's processing modules as 'component' plug-ins for separate use within your host DAW software. Pictured here is the Pitch module.

» However, if you want more direct control, the Pencil button lets you customise the number of harmonies and their respective pitches, while the Mixer button (the icon with two semicircles) provides a visual mixer that lets you arrange the stereo positions and levels of the various harmonies. If you prefer to 'play' your harmony parts, the MIDI button lets you voice them from your MIDI keyboard too.

The new Backer module does something slightly different. It takes an original vocal line and transforms it to sing the same line (in unison, or up/down a whole octave) with a different voice. Out of the box, eight voice 'persona' presets are included, providing a mixture of male and female voice characters. You also get options to shift the tone and formant properties and adjust the blend between the real and synthesized voice. When you have recorded a single singer as the source of all your vocal layers and harmonies, Backer therefore allows you to change the character of some or all of the additional vocal parts, creating much more of a 'vocal group' effect. It works remarkably well and, interestingly, you can also create your own custom personas by letting the module analyse an audio file containing the target voice. I tried this with a few isolated vocals from commercial tracks and, although you can't, for example, turn your own vocals into a captured version of Adele or Chris Cornell, it definitely lets you apply a different tonal flavour. It will be interesting to see where iZotope take this technology in the future and whether, eventually, it will be possible to get Nectar's Voice module to use Backer's personas for its auto-generated harmonies.

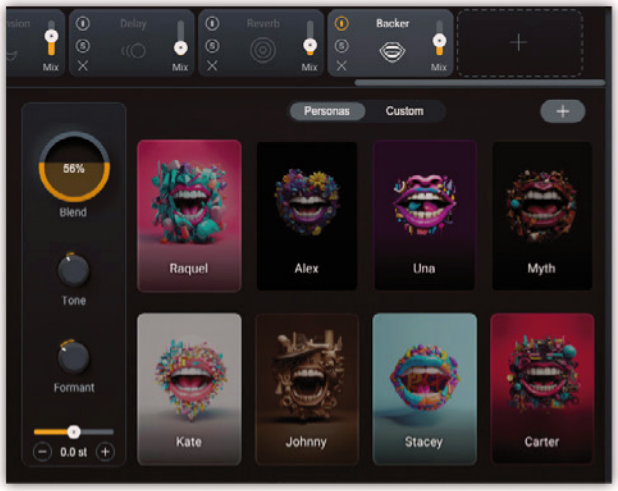
Within the Voices module, Nectar 4's revamped auto-harmony features are both flexible and easy to use.

Release The Plug-ins!

In the Detailed View of Nectar, the list of processing modules available remains impressive. You get the Compressor and EQ (two instances of each are available), Backer, De-esser, Delay, Dimension, Gate, Reverb, Saturation and Voices. But the Advanced edition now also allows you to access many of these modules as individual plug-ins (which iZotope refer to as 'component plug-ins') in your DAW. This is a most welcome development, as it means they can now be used instead of, or in combination with, your stock or third-party plug-ins. Incidentally, Nectar Standard and Advanced users now get automatic pitch correction, and for Advanced users this is a component plug-in too. While we're on the subject of pitch correction, it's also worth pointing out that a copy of Celemony's Melodyne 5 Essential Edition is bundled along with the Standard and Advanced editions.

Ready Mixed

Nectar 4's new and improved features represent a significant step forward from earlier versions, and there is something



Backer's UI might have a cartoon feel to it but don't let that fool you — the results can be very convincing indeed.

useful for both new and experienced users alike. Perhaps unsurprisingly, therefore, the asking price for Nectar 4 Advanced is not insignificant. But that's not to say you don't get great bang for that buck — you do. And it's hard to deny that NI's Music Production Suite 6, which includes Nectar 4 Advanced, Ozone 11 Advanced, Neutron 4, RX 10 Standard, Guitar Rig Pro 7, Tonal Balance Control 2, Neoverb and VocalSynth 2 amongst others, is even better value.

So who might Nectar 4 appeal to most? Well, the AI assistance and the presence of the Voices and Backer

> **"It is, frankly, incredibly easy to create harmony parts that, sat at a suitable level in your overall mix, will be genuinely convincing."**

modules in the more accessibly priced Elements edition should certainly make it attractive to those who are new to Nectar, and perhaps have less audio engineering experience under their belt. The Advanced edition is obviously at the other end of the price range, and if you're looking at this, you are probably considering Nectar 4 for its serious processing capabilities.

The plug-in offers a tremendous level of control if you want or need it, allowing you to craft and refine your vocal recordings in both surgical and creative fashions. However, perhaps the most impressive feature of Nectar 4 is how quickly it lets almost anyone go from a raw vocal to a 'produced' and mix-ready voice. Whatever your level of experience, having Nectar's AI give you a head start with a processing chain is a huge time-saver. In an AI instant, a dollop of Nectar is an easy, and very effective, means to sweeten your vocals.

$ $299. Discounts available for upgrades from previous versions. Also included in Native Instruments Music Production Suite 6 $599.
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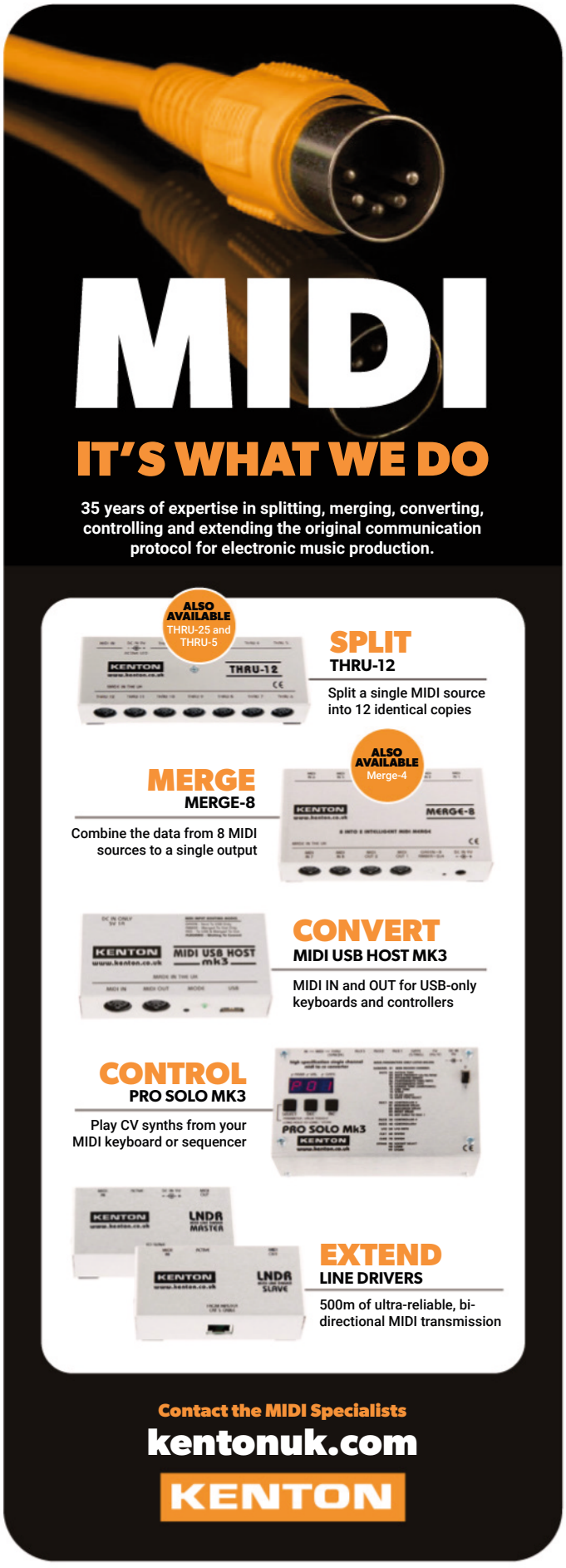


Cherry Audio Pro Soloist Software Synthesizer

Cherry Audio have resurrected the venerable ARP Pro Soloist in software.

GORDON REID

I make no bones about my love for the ARP Pro Soloist, and I wouldn't be surprised if there were secret cabals in *SOS*-land who regularly take the piss out of me for referring to it in reviews of other equipment. I don't care. Every time that Korg have hinted at another recreation of a vintage ARP, or Arturia have teased their next slew of soft synths, my fingers and toes have been crossed in the hope of an accurate recreation of this funny little preset monosynth. And, finally, it's arrived. But it's come from elsewhere.

You might be wondering what made the Pro Soloist special, and why I think that it remains so. To answer this, let's return to 1971, a year in which changing from one synthesized sound to another was an activity that could at best be described as hit-or-miss. Most of the time it involved repatching, although, on the new-fangled Minimoog that was just about to appear, a pre-defined signal path meant that you needed only press a few switches and twist a bunch of knobs to obtain something that — if you were lucky — approximated the sound you wanted. But this was also the year in which ARP released the Soloist, a small synth that allowed you to jump from any of its 18 preset voices to any other at the flick of a toggle. Not only did this sound great, it introduced something revolutionary that we now take for granted. Despite Cherry's website and manual claiming otherwise, the Soloist boasted aftertouch with up to six simultaneous destinations: pitch, vibrato, wow, growl, brilliance and volume.

The Soloist was a marvel but, sadly, it was flimsy and prone to failure. So, the following year, ARP release the Pro Soloist. This also had its shortcomings, but it exuded one quality that overrode any criticisms: its 30 presets could sound amazing. But how? Surely, it's just a single-oscillator preset monosynth with a signal path comprising nothing more than a VCO, VCF and VCA, right? No. Wrong. Very wrong. In many ways the Pro Soloist was the most innovative of all ARP's synthesizers, and much of its architecture was unique, including a digital keyboard scanning system and a fascinating audio generator that comprised a high-frequency oscillator driving a logic counter, which in turn produced a stream of pulses that were used to generate the initial waveforms. Then there was the way that ROM chips were used as switch matrices to change the voicing when you selected a new preset. Furthermore, there were multiple signal paths through the synth, one of

Cherry Audio Pro Soloist

$49

PROS

- It really sounds like a Pro Soloist.
- Having access to the underlying voicing makes it far more flexible than the original.
- The extended 16-voice polyphony and bi-timbral architecture take it where no ARP ever went before.
- It's eminently affordable.

CONS

- The simple single-oscillator architecture means that some types of sound can't be obtained.

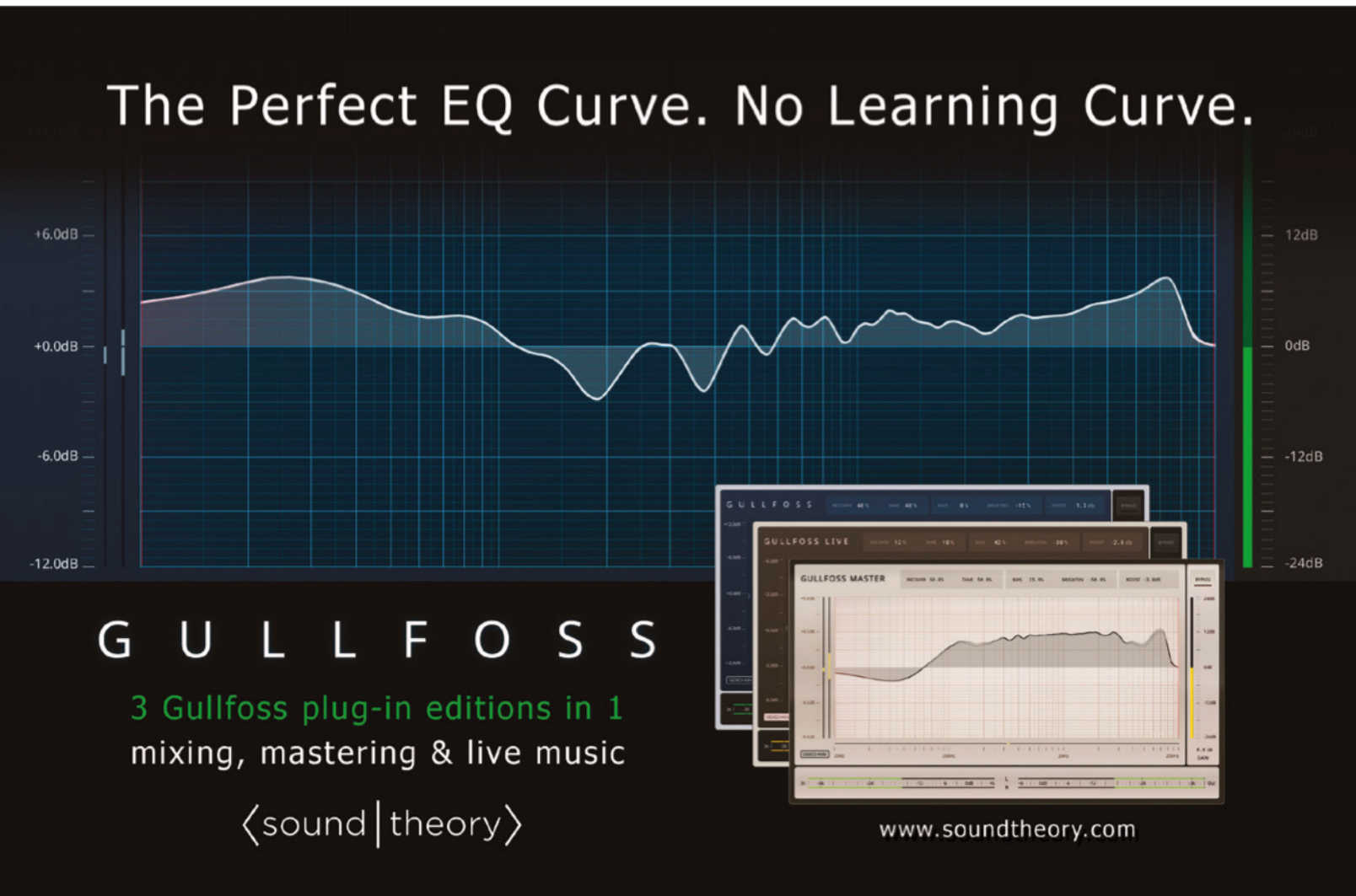
SUMMARY

I have been impressed by some recent soft synths but, even in the studio, I almost always turn to something that hurts when you drop it on your foot. So if I tell you that this is one of the few softies that I might use in preference to its inspiration, I don't think that I need say more.

which passed the oscillator's output to three filter banks containing 10 tuned resonators. When you selected a voice, a bunch of digital gates directed the signal through (up to) five of these resonators to obtain complex timbres that, even today, can be produced on little short of a modular synth or its software equivalent. Not only could the Pro Soloist produce orchestral sounds that were more accurate than anything else of its era, its aftertouch made it one of the most expressive synths ever produced, and some of its sounds remain revered to this day, especially amongst fans of mid-'70s Genesis. The Pro Soloist was itself superseded by the Pro/DGX in 1977, but later versions of this replaced the revered ARP4012 filter of the Soloist and ARP4034 of the Pro Soloist and early Pro/DGXs with the ARP4075, and were considered (perhaps rightly) as inferior, so we'll say no more about it.

Standing On The Shoulders Of Giants

The Performance screen of Cherry Audio's Pro Soloist looks much like the original synth, although a closer inspection reveals that it's a hybrid of the Pro Soloist and the Pro/DGX and offers several new features including aftertouch-controlled wah-wah and tremolo. Since the GUI is too small to contain everything that the soft synth offers, a second screen provides access to all of the hidden goodies of the Pro Soloist and then adds many more. For example, you can now program the sync'able LFO and both contour generators, access all of the wave generation in the oscillator (which now offers a Super Wave option), and determine how much of which waveform (or waveforms) passes down each of the audio paths. You can also program the all-important filter emulation, and this now boasts an unusual feature: you can change the polarity of its output to create frequency cancellations and reinforcements when you mix it with the other audio paths through the synth, and this greatly extends the sound design potential. In addition, Cherry have replaced the original resonator banks with five programmable resonators rather than 10 fixed ones. Happily, each of the original curves is available as a preset, thus making it possible to obtain any of the original voices. There's also a six-slot modulation matrix. Each slot contains a source, a 'via' »



» and a destination. (A via allows you to control one modulation source using another.) There are 22 sources, 22 vias, and 57 destinations on offer and, since most things can be controlled using Cherry's usual MIDI Learn system, it's immensely powerful.

The third screen allows you to polish your sounds with five dedicated effects sections that can be sync'ed to master clock and MIDI Clock where appropriate. Distortion offers a choice of tube and fuzz modes; the phaser offers a choice of poles for a range of famous phasing effects; the flanger/chorus offers a selection of modulation waveforms; the echo has digital, tape and ping-pong modes plus stereo spread; and the reverb offers five modes with high-pass and low-pass filters to sculpt the reverberated signal. The results are not as whooshy (or as noisy) as passing the output from an original Pro Soloist through a Big Muff, Phase 100, Electric Mistress and Space Echo, but they extend the capabilities of the soft synth without tying up your plug-ins or external boxes, and that's no bad thing. The final feature is a sync'able arpeggiator with up to three octaves' range and up, down, up/down and random modes.

The Sound Of History

Although I was nowhere near my studio when Cherry Audio sent me the soft synth, I couldn't resist launching it, selecting the Tuba, pushing it up an octave and... Oh my word! Even when played from my computer keyboard and monitored through headphones, it farted at me in exactly the way that it should. This was a great start! Later that evening, I liberated a Pro Soloist from storage and compared the preset sounds of the soft synth with those of the original. Did I expect them to be identical? No, I didn't, and neither should you, because not all Pro Soloists sound the same. If the components in one unit differ from those in another so that (for example) the centre frequency of a given resonator strays by a few percent, the two sound slightly different, because there's nothing that you can do to bring them into line with one another. So was I concerned by the small differences that I discovered when comparing the soft synth with my Pro Soloist? Not at all. On the contrary, I was impressed. I sometimes had to make small adjustments in the soft synth to match the two but, as often as not, all that was needed was a tweak of the Brilliance to make the two functionally identical.

"My favourite brass sound of the '70s was the Pro Soloist's Tuba transposed up an octave with Growl, Brilliance and Volume controlled by aftertouch. Again, the soft synth recreated this perfectly."

The soft synth retains and extends the Pro Soloist's pressure-sensitive capabilities, now offering channel and poly aftertouch as well as an innovative 'last note' mode that, when receiving channel aftertouch, controls the modulation of just the last note played. This makes it possible to emulate some of the things that you might wish to do with polyphonic aftertouch even if you don't have a suitable controller to hand. I tested all of this, and was again impressed. I have always loved the Flute preset with the Volume and Vibrato controlled by aftertouch, and the soft synth recreated it with ease. Likewise my favourite brass sound of the '70s was the Pro Soloist's Tuba transposed up an octave with Growl, Brilliance and Volume controlled by aftertouch. Again, the soft synth recreated this perfectly. By this time, I was confident that all of the other voices would respond much as expected, and so it proved.

It was now time to play the soft synth in anger, so I set it up to respond to my playing in the way that I wanted. When using it as a monosynth, you can select between low-, high- and last-note priority with a choice of single- or multi-triggering. Alternatively, you can turn it into a polysynth with a maximum of 16 voices per layer. Hang on a sec... Polyphony? Per layer?

Everything that I've described so far is but half of the soft synth because it offers Upper and Lower layers that duplicate everything: sound generation, performance settings, MIDI control, effects and arpeggiation. You can split and layer these to create two independent monosynths or two independent polysynths, or one mono and one poly, or bi-timbral sounds, the last of which allows you to create some huge basses and lead-synth patches. When programming, you can Solo either layer in place to check that it's doing what you think that it should, and a small utility menu allows you to do useful things such as copying, swapping, importing and exporting. The only thing that's global when you use it in this way is the limiter found on the FX screen, which ensures that you don't distort one sound by pushing the other too hard. Hmm... that's

Unlike the original, Cherry Audio's Pro Soloist gives you full control over the synth from the Edit page.

The third screen is dedicated to the built-in effects and an arpeggiator.

not quite true; if you want to weld two disparate sounds into a coherent whole, you can select a Global FX mode that causes both layers to pass down a single effects path to help glue them together.

So how good is a Pro Soloist as a bi-timbral polysynth? I must admit that — yet again — I was impressed. Monosynths can sometimes be overpowering when turned into polysynths because there's too much heft behind each note, but the Pro Soloist survives the transition to polyphony in rude health, and I found it simple to program warm strings, pads, brass ensembles and choirs. It's not a synth you'd choose for things such as chimes and pianos, because the lack of oscillator sync and FM largely precludes these but, in contrast, I was surprised by the ease with which I could obtain the floaty Vangelis-esque sounds that pour out of it. It's also capable of a surprising range of off-the-wall sounds and effects, courtesy of the modulation matrix. Nonetheless, the Pro Soloist's forté remains orchestral sounds and, whether my target was a solo flute above an *Oxygene*-inspired string accompaniment, or brass solos and stabs, or dual-layered ensembles, the results were almost always superb.

Before finishing, I tried performing live (admittedly, with no-one watching) with the soft synth. Controlling it from the GUI is clunky but I soon found that, in addition to the expected MIDI control and automation, I could assign each voice tab to a fader on my MIDI controller and then move the appropriate one up or down to select the upper or lower voices depicted on the GUI. Once I had programmed this, I was then able to recreate the solo from 'The Cinema Show'. This is perhaps the definitive demonstration of the Pro Soloist, and famous for the way that Tony Banks flicks between presets as he plays. (OK, that doesn't sound particularly impressive in 2024, but it was gob-smacking in 1973.) Since MIDI maps can be programmed on a per-patch basis, it's even possible to allocate different presets to different faders for different tracks, which makes things even better for live performance. For the fourth time, I was impressed.

Finally, I should also note that all of the housekeeping (patch management, GUI settings, audio and MIDI setup, oversampling settings, and so on) is common with other Cherry soft synths, so I hope that you won't mind that I've skipped these here.

Conclusions

Pro Soloists are rare, and those that are still fully functional are even rarer. Nevertheless, the character of these unassuming little monosynths means that their reputation has survived for more than half a century — and not just among keyboard players in Genesis tribute bands. Cherry's Pro Soloist mimics them with remarkable accuracy but, whereas the originals are constrained by their preset nature, the extended capabilities of the soft synth take it far beyond anything envisaged by ARP's engineers in 1972. Well done, Cherry Audio.

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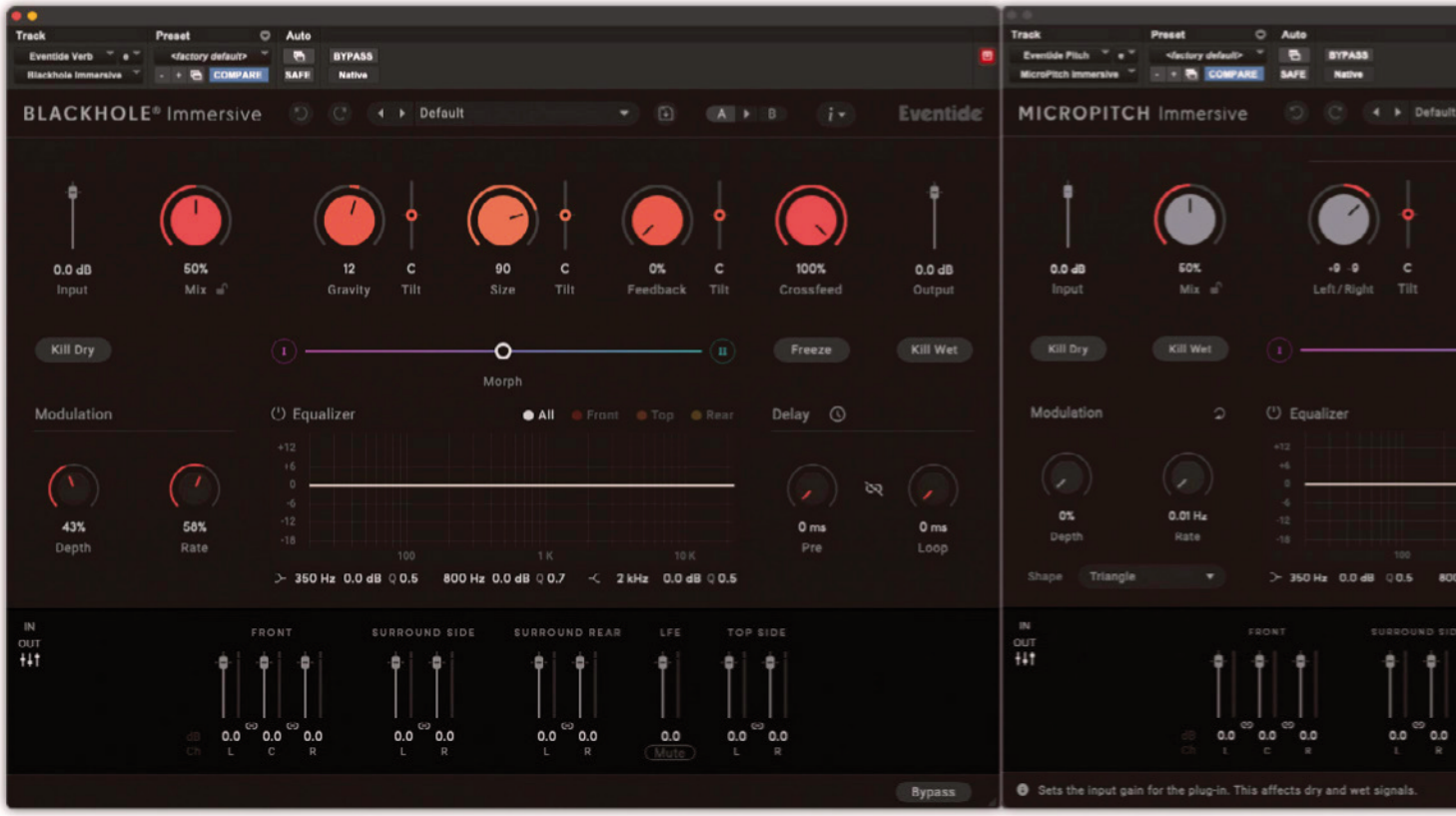


Eventide Blackhole & MicroPitch Immersive

Multi-channel Effects Plug-ins

Eventide's celebrated reverb and pitch/modulation plug-ins have been reinvented — with immersive audio applications in mind.

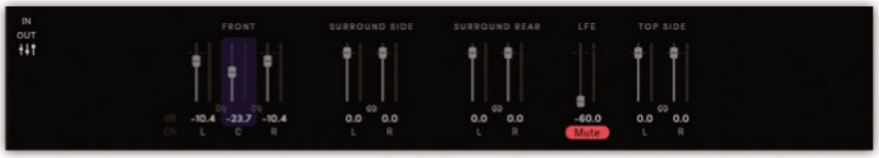
TREVOR MICHAEL

Way back in *SOS* September 2012, Paul White reviewed the original Eventide Blackhole reverb plug-in; more recently, in August 2020, he assessed their MicroPitch. Towards the end of last year, Eventide released new versions of both plug-ins with the multi-channel capability needed by those working in immersive audio. The plug-ins come in AAX, AU and VST3 64-bit formats, for Windows and Mac operating systems. Windows users will need Windows 8 or higher, Mac users OS 10.14 or higher, on either Intel or Silicon machines, and you'll also need an iLok account. The original stereo-only versions are still available separately (and more affordably), and if you already own those, Eventide offer a more affordable 'crossgrade' option to the immersive versions.

Overview

Both the new plug-ins are laid out in similar fashion, with the same colour scheme and corresponding controls in the same locations. Those accustomed to the stereo versions will see a lot of familiar controls here, but there are obviously quite a few differences, too — not to mention a bunch of new presets. The multi-channel output formats supported are LCR, Quadraphonic, 5.0 to 5.1.4, and 7.0 to 7.1.4; following an update in January 2024, mono, stereo and 9.1.6 are now supported too. I was initially unable to get the 7.1.4 option working on Mac/Pro Tools Ultimate 2023.6, but happily the update fixed this. The Immersive plug-ins will upmix to whichever of those layouts you desire, so you can insert an instance on a mono or stereo channel and route the channel output to a quad bus or 7.1.2 bed, for example.

The plug-in displays can be resized by dragging from the bottom right corner, and holding down the Command key on Mac or Ctrl key on Windows while changing parameters lets you fine-tune the settings. Several controls and parameters are common to both plug-ins too. The input and output level are each adjustable between -60 and +12 dB, and along the bottom of each plug-in are input and output



The mixer, available in both plug-ins, allows you to determine how much of the effect output is sent to the different channels in your immersive mix project.

meters, displayed as simple dots, to give you an indication of when signal is present, or if you're overloading the ins or outs anywhere. Clicking on the mixer icon in this section gives you the option to control the level of the effect going to each output channel. For example, on a 7.1.2 instance you're given output level control over the front LCR channels, L and R sides, L and R rear, LFE (which also contains a mute option) and L and R top.

In the centre of each GUI is a very flexible three-band EQ, with high and low shelves (both switchable to cut filters) and a parametric mid band. This can be used as a general EQ, over the whole effect, but the front, top and rear channels can also be EQ'd independently. Mix is obviously a control that sets the blend of the dry and wet signals. This has a useful lock function, which ensures that your chosen mix blend will remain the same when auditioning different presets. It may seem a simple idea, but it's so helpful! The Kill Dry button mutes the input signal, and this can be especially useful on the Blackhole reverb plug-in, as it makes it easier to audition and tweak the reverb sound when you insert this plug-in on an audio track rather than using it as a send effect. Kill Wet is a momentary button that damps the reverb tail and mutes the effect. Both buttons can be automated, which opens the door to some clever creative effects.

Though it's been renamed Morph, the ribbon control from the stereo version is still here, and operates in exactly the same way. You start by hitting the 'I' button on the left of the slider and then set up your sound. Next, hit the 'II' button to the right of the slider and adjust some settings. The slider can now be automated to sweep between the two 'presets' you've just set up. It's a lovely feature that, I suppose, offers a similar approach to that which an expression pedal might for a guitar effects processor. It makes automating plug-in settings a lot quicker than tweaking each parameter one by one.

All in all, between the EQ and output level controls, there's all sorts of possibilities available to tailor the effect exactly how you want it in the immersive environment: wetter at the back, dryer at the front, brighter at the top… whatever you need.

Blackhole

Blackhole is a beautiful, luscious, dark-sounding ambient reverb. The algorithm began its life in the popular Eventide DSP4000 hardware effects unit back in the early '90s. Owing to its popularity with guitarists, it was released as a dedicated guitar pedal, before also appearing in the Space Guitar and then H9 pedals. But for those of us who are mixing in DAW software, the plug-in versions are very convenient!

As mentioned previously, those familiar with the stereo plug-in versions will see a lot of the same parameters here, such as Gravity, Size, and Feedback, which all affect decay time in various ways. Gravity alters the shape of the decay. At the 12 o'clock position, the decay is dense, and it gets smoother as you turn the knob clockwise. Go anticlockwise from the same point and you're into reverse-reverb territory. Pulling back the Size parameter can give tight slap-back-like effects, and increasing it takes things to canyon-esque proportions, while Feedback obviously increases the tail of your reverb — but it can do so all the way to infinity if desired.

Crossfeed is a new parameter that doesn't appear on the stereo version, and this controls how much of the reverb effect is fed into the surround channels. At its minimum setting, the reverb effect appears only on the channels coming into the plug-in. For example, a 7.1.2 instance of the plug-in fed from a mono source with Crossfeed at 0% will produce a mono output in the centre channel. Crank the Crossfeed up and the reverb will start feeding all the other channels.

The tempo sync of the stereo Blackhole plug-in has been treated to an upgrade in the delay section. We can set the pre-delay here manually from 0 to 2000ms, or can sync the delay to the DAW tempo in beat subdivisions. The loop delay function sets the amount of delay after the reverb function and feeds back into the reverb, creating huge ambient textures.

Another difference between the stereo and immersive versions of both plug-ins lies in the handy Tilt controls, which weight the effect of parameters towards the front or rear of the room. On Blackhole, these are available for the Gravity, Size and Feedback controls. Basically, you can just set the parameter value to the maximum you want. Tilting the parameter towards the front speakers will then reduce the parameter at the rear, or you can tilt towards the rear of the room to reduce the parameter at the front. Leaving these »

Eventide Blackhole & MicroPitch Immersive

$499 each

PROS

- First and foremost, great-sounding algorithms.
- Can do subtle or out-there effects.
- Versatile multi-channel routing.
- Some clever immersive-oriented controls.

CONS

- Priced for the pro market!

SUMMARY

These plug-ins do more than simply up the channel count of their stereo equivalents: a range of new features makes them relevant to immersive audio professionals.

The EQ section on both plug-ins can operate globally or on individual channels.

A ribbon controller, now called Morph, blends between two sets of parameters — and can make automation of multiple parameters easier.

» settings in the centre position will ensure the same values front and rear.

There's a modulation section to set the depth and speed of modulation in the reverb tail and, finally, the Freeze button. When engaged, this mutes the dry input signal and gives instant infinite reverb. Automate some parameters when freeze is active and you can have all sorts of fun. It's very cool.

MicroPitch

What Eventide don't know about modulation and pitch processing probably isn't worth knowing, and their processors have long been used to give mono sources such as guitars or vocals a sense of 'spread'. So it makes perfect sense that these algorithms would find a home in immersive audio productions. But while Blackhole Immersive is, in terms of functionality, broadly similar to the stereo version, the MicroPitch Immersive plug-in is quite a different beast to its stereo counterpart.

Eventide tell us that they "designed MicroPitch Immersive from the ground up for Immersive, Dolby Atmos, and surround workflows. With up to 12 channels of spatialised, detuned delays, plus modulation, EQ, and crossfeed controls, MircoPitch Immersive excels in creating an immersive space with boundaries that reflect, move, diffuse, and dissolve. Whether it be adding a touch of space or creating delayed, pitch diving chaos, MicroPitch Immersive will help you architect both real and unreal immersive realities."

The main controls for the pitch-shifting are found in the Detune section. Here, three knobs each look after a pair of parameters, namely Left/Right, Front/Rear and Top/Main. At their default centre positions, no pitch change is applied. Moving the control clockwise on a pair shifts the former parameter's pitch up and the latter's pitch down, by between zero and 50 cents. For instance, if we take the Front/Rear control and turn it clockwise, the Front pitch increases and the Rear pitch decreases by the same amount. Moving the control anti-clockwise decreases the Front pitch and increases the Rear.

Each of the three detune controls also has the Tilt function I discussed above when describing Blackhole. So, again, you can set the pitch-shift on each of the three controls to the maximum you want, then tilt as desired to offset the value between Left/Right, Front/Rear or Top/Main. The Feedback and Crossfeed controls also operate in the same manner as for the Blackhole plug-in, and the delay settings are the same too (adjustable manually from 0 to 2000 ms, and can be locked to the DAW tempo). An additional drop-down menu in this section is the delay pattern option, which makes MicroPitch a very useful multi-channel delay plug-in (quite apart from its pitch processing). There are three choices: Self, whereby the delays will appear in their normal position, or Rotate Left or Rotate Right, which make the delays rotate around the room in the chosen direction. It's not something you'll want to do every day, but then again, why not? It's early days in immersive audio mixing so we should experiment!

Finally, the modulation section has a 'shape' drop-down that's not in the stereo version, and this provides Triangle, Square, Saw up, Sine, Saw down and Random waveforms.

In Session

I used both plug-ins when mixing a live album in Atmos, the project being based on stereo stems provided by the producer and stereo mix engineer. My aim was to faithfully capture the vibe and intent of the stereo mixes, yet also put the immersive listener right at the gig, in the venue.

On this particular album, Blackhole was useful on electric guitars and pianos. The reverb from the stereo mix had already been printed into the stems, but I was still able to send a little to a 7.1.2 instance of Blackhole, using the output gain controls to take the effect out of the front LCR speakers, thus adding reverb only to the sides, rear and top. This succeeded in creating the sense of an immersive space, but with all the main energy still coming from the front speakers. More ambient-sounding guitars benefited from a longer Blackhole reverb, with a little Tilt to lessen the effect at the front of the room, where the guitars were placed, and enough in the rears to give a sense of the sound being 'around you'. I found that with these more ambient and dense textures, the mix remained clear and uncluttered, even when checking the folded-down binaural and Apple spatial audio versions in headphones. I ended up using MicroPitch on a stereo keyboard pad. It subtly spread the keys around the room in such a way that you almost didn't realise it was doing anything until you took it away. So it was an unobtrusive effect, but it still served to pull the listener deeper into the immersive experience.

Conclusion

Blackhole is an awesome-sounding, dense-yet-transparent ambient reverb, and the immersive version lets you tailor the sound in incredible detail across the entire multi-channel array. It's more CPU-hungry than many other multi-channel reverb plug-ins, but if you want huge, lush, ambient soundscapes this plug-in does it better than most. As you'd expect, given Eventide's decades of experience in pitch and delay processing, MicroPitch Immersive is pure quality. It can be subtle or produce more in-your-face creative pitch changes or modulation effects, and tight panning delays in the Immersive world. If you work in immersive music, these will be very desirable tools!

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NOMINEE
SOS AWARDS 2024

SOS EDITORS' PICK
GEAR OF THE YEAR 2023

Two notes Genome

One of Genome's amp models: you can tweak the power amp settings, including the valve types and configuration.

BOB THOMAS

Amp, Cab & Effects Modelling Plug-in

Combining Two notes' highly respected models and dynamic IR-based cabinets with support for third-party machine-learning amp captures, Genome is brimming with potential.

Launched by Two notes in January 2024, Genome is described as a "carrier-class software ecosystem engineered to deliver the pinnacle in end-to-end tone shaping". What these words boil down to is an amp, cab, pedal and effects modelling plug-in that features not only Two notes' modelled amps, effects and cabinets, and their now familiar dynamic impulse response engine, but also a new CODEX "next generation unification engine" that supports third-party AI models from Neural Amp Modeller, Aida-X and Guitar ML. Genome is compatible with VST, VST3, AU and AAX hosts and requires, as a minimum, a Windows 10 or macOS 10.15 (Catalina) computer.

Globalism

Before we dig in to the sound and feel of this software, I need to cover some of the terminology, the main GUI and the global controls that apply whatever signal chains are currently loaded.

Two notes refer to a chain of amps, cabs, pedals and studio effects as a 'lane', and a lane can be saved and recalled as a 'rig'. A lane is made up of blocks called 'inserts', that are always present, even if empty, and display the category of the loaded model. Hovering the cursor over an insert displays the model name, and a click opens the model up for editing.

Running horizontally across the top of Genome's GUI is a two-level strip of common controls and facilities. A setup menu allows you to sign into your account (unless you activated Genome offline, you need to be signed in to use it), and perform other admin tasks. Next to that sit a stereo/mono input configuration selector, an input level fader and metering, a large-format tuner, and a well-featured noise gate. On the right, the output level fader and its metering are joined by a CPU meter.

Above this, a drop-down menu accesses oversampling options. As usual, increasing the internal sample rate reduces aliasing at the expense of greater CPU usage, but the Ultra

Two notes Genome
$80

PROS
- Can combine AI, modelled and impulse-based emulations in one signal chain.
- Great-sounding models of amps, cabs, pedals and studio effects, with deep editing capabilities.
- Can host AI captures from NAM, Aida-X and Guitar ML's Proteus, and 42 great captures are pre-installed.
- Very competitively priced.

CONS
- The noise gate can only be applied globally.

SUMMARY

With quality amp, cab and effects models, extensive editing capabilities, and the ability to host or integrate the AI-capture formats of three leading open-source AI modelling companies into a signal chain, Genome is an impressive plug-in for a great price.

and Max settings are intended for offline rendering, delivering a slight improvement in quality at the expense of very high CPU use.

On the lower left, a text box names the loaded rig and has navigation arrows to move up and down the list, or can display the entire list. Rigs are organised in eight banks of 128 (0-127) for a maximum of 1024 rigs, the first 222 being populated with factory presets. Icons for the save, undo, redo and revert (to saved rig) functions are adjacent to this, and on the far right, a drop-down Automation menu accesses 10 user-configurable DAW control automation assignments, which can be stored within a rig. Another 14 hard-coded DAW assignments cover bypass, input gain, output level, noise gate parameters, and lane level, pan, mute and phase. The final global feature is a tempo display, with options to use the tempo saved in the current rig, tap tempo, or sync to the host DAW.

Parallel Lanes

Genome's audio input to can be set to stereo, mono L-R mix, mono L or mono R. This signal feeds a lane that runs left to right and contains 10 inserts, each of which can be loaded with any model, from any category and in any order. To create a parallel path, you can place a split tool (one instance only) before any insert, and a merge tool (with a mixer to control lane levels and pans) after any insert. For example, placing thea split before the first insert and the merge after the last results in two parallel lanes, with 20 inserts available in total. Reordering a lane is easy, as loaded inserts can be dragged to replace empty ones. The split and merge can be moved in a similar way.

Some of the terminology in the online user manual is inconsistent, so a little confusing: it variously refers to Genome models as "components", "component modules", "models" and "effects". But In practice, it's fairly intuitive. A left-click on an empty insert opens up the Add Effect menu, which lists 11 'component' categories. A right-click opens the Add Effect menu's evil twin, Replace Effect, with which you can bypass, delete or replace the loaded 'effect'. Each inserted 'effect' has a ±12dB output level trim control, labelled FX Out.

The list of modelled amps features the same Tube Stage Modelling (TSM) pre and power-amp models as found in Two notes' Opus pedal (reviewed in *SOS* December 2023: https://sosm.ag/two-notes-opus), plus a TSM power-amp model that allows external hardware preamps to take advantage of Genome's power-stage options. All TSM amps feature Depth and Contour controls, the former adding low-frequency content to the sound, and the latter changing the level of bass and treble frequencies to shift the overall sound of a model between a mid-focused and a scooped-mid tonality. TSM amp power stages can be switched between 6L6, EL34, EL84 or KT88 output valves, single-ended Class A or push-pull Class A/B, and triode or pentode mode.

Genome also offers 12 modelled effects pedals. Alongside an MXR Carbon Copy delay, MXR Phase 90, EHX Big Muff Pi, Ibanez TS-9 Tube Screamer, Klon Centaur, first-gen Marshall Bluesbreaker, ProCo Rat, BOSS SD-1 Super Overdrive and DOD Overdrive Preamp 250, you get separate bass- and guitar-oriented seven-band EQs and a volume pedal with heel/toe scaling. There are also 'studio' effects, which tend to be more tweakable: a dual-channel delay, a decently featured stereo reverb, chorus, vibrato, tremolo, phaser and flanger, an EQ with up to 16 bands and a familiar node-on-curve interface, a compressor with gain-reduction history, an enhancer with separate settings for guitar and bass, an ADT effect, and a harmonic enhancer.

Clever CODEX

So there's already a hell of a lot included in Genome — and then there's CODEX, selected from the amplifier category. Clicking anywhere in the model-name field of the CODEX GUI brings up a list of 42 pre-installed NAM-, Aida-X- and Proteus-format amplifier captures. If you »

Two notes' proven DynIR cabinet emulations are included, and allow you to select the mic types, distances and positions.

What really sets Genome apart from the crowd is the CODEX engine, which allows Genome to host and manipulate third-party AI-based amp captures.

want to use your own or other third-party AI model captures, you can drag and drop them into the Captures folder that Genome creates on installation, or load them from another folder. You can scroll sequentially (in alphabetical order) through the list of AI models.

At the time of writing, CODEX doesn't support sub-folders in the Captures folder so, if you value your sanity, I recommend keeping your AI model downloads organised by type in another folder. Also worth mentioning is that a consequence of CODEX being in the Amplifier category is that all CODEX inserts are labelled as amps — even if they're pedals or studio effects! These aren't insurmountable problems, though, and could no doubt be addressed in a future update.

Once an AI-captured model is loaded, you can warm or brighten the input signal, apply gain to it, and/or run it through a bypassable tone stack. Fender and Mesa Boogie-style tone stack models are the only two currently available, but more are planned. The Pre/Post control on the right of the tone stack switches CODEX's internal signal routing between Input Signal/Tonestack/Gain/Input Voicing/Model/Output/EQ/Enhancer (Pre), and the more traditional Input Signal/Gain/Input Voicing/Model/Tonestack/Output/EQ/Enhancer (Post). The modelled input signal then enters CODEX's five-band graphic EQ, which is topped and tailed by low- and high-pass filters (2-10 kHz and 10-500 Hz, respectively). The centre frequencies have presets optimised for guitar and for bass, and a custom option is provided for those whose instruments require more tailored control.

The equalised signal feeds the enhancer, which is a kind of three-band dynamics processor in which applying compression to individual bands can help you sculpt the final tonal profile. Its three EQ bands (Thickness, Body and Brilliance — in other words, low, mid and high) control the amount of frequency processing, while a dry/wet blend control determines the overall amount of 'enhancement' being applied. Finally, there's a master output level knob. After you've adjusted a model, you can save the settings as a CODEX preset. (An important point is that all the Two notes models — not just the CODEX ones — have this facility, so you can amass presets of your favourite settings within every Two notes model).

Dyn IR & IR Loader

No Two notes review would be complete without mentioning their DynIR cabinet emulations and Torpedo Wall Of Sound (WOS) plug-in. The WOS plug-in has two components: a DynIR Manager/Loader and the Torpedo Post Effects. The latter gives you a combined EQ, exciter, compressor and reverb that you can use anywhere in a lane, and there are 13 (10 guitar and three bass) DynIR cabinet emulations that can be configured very precisely. As well as loading a DynIR model, WOS allows you to change the two selected mics. In addition, you can change the mics' distances from the virtual cab, their positions on the horizontal axis, their phase relationship, the amount of distortion, and their absolute and relative volumes.

All this can be saved as a DynIR preset, and DynIR presets created in earlier versions of WOS are supported (though only the first two of a possible six cabinets will be loaded). Adding icing to this cake, the TSM power amp can be inserted in a mode that replicates precisely the behaviour of the power amplifier section of the legacy WOS plug-in. Genome's IR loader allows you to load two third-party IRs, and to change their phase relationship, amount of distortion, and absolute and relative volume levels.

The Bottom Line

So, that's what you get, but just how good is Genome? Two notes' traditionally modelled amps, pedals and studio effects compare extremely well with the best competition in terms of sound quality. These amps may lack the range

of adjustment options that help me coax very convincing emulations out of similar models in Bias Amp 2 (my go-to in this area) but they sound every bit as good, and the effects models hold their own very well against the equivalents in Bias FX2. The DynIR cabinets are extremely good too, again bearing comparison with the best. But if you've used Two notes products recently, you'll know all this already.

What sets Genome apart from the crowd is its support for AI machine-learning amp/cab/effects captures. As in so many sectors, AI technology is disrupting the software amp-modelling market. Its impact is due not only to the accuracy of the models this approach can deliver (generally, I've found that when it comes to recreating an amp sound, a good AI capture will walk all over an equivalent model created in more traditional ways) but also to the open-source development ethos of platforms such as NAM, Aida-X and Guitar ML (Proteus). In that respect, it's somewhat reminiscent of what happened when sampled instruments or impulse response capture software first started to emerge.

Choice, though, can a double-edged sword. On the one hand, you'll almost certainly find something you want if you spend time searching and auditioning the different options. On the other, you might have to work pretty hard to find the best versions, as there are literally thousands of free models available on the Internet, ranging from full rigs to the individual elements in the signal chain. It's great, then, that Genome comes with a curated collection of 42 AI models. These all deliver superb facsimiles, including several of my favourite amps and pedals, and their performance more than justifies the excitement currently raging around AI modelling. It can also show those of you who haven't yet got into this technology just how good AI models can be. By way of example, there are both traditional and AI models of a Klon Centaur in Genome, and the AI version sounds much more detailed to my ears. Similarly, there's a modelled Fender amp called Foundry and an AI-modelled version of the same amp. While the AI model does sound very like the Foundry, there's just more detail in the sound. It's a little more open: more alive, more accurate.

While it's great that Genome supports all this, though, it also goes a very helpful step further. Some of the more traditional models sound as great as they ever did — we shouldn't write these off just because there's a new tech in town — and the DynIR approach delivers some very convincing cabinets. The freedom that Genome's CODEX engine gives us to combine and manipulate elements created using all these different technologies in a single signal chain, inside a single plug-in — one that, importantly, isn't tied to any hardware — is Genome's trump card.

As well as a good selection of modelled stompboxes, Genome features a range of more tweakable 'studio' effects to help you shape your tone.

ALTERNATIVES

There are plenty of capable amp modellers from the likes of **UA**, **Positive Grid** and **Line 6**, but that's only a part of what **Genome** does. The closest competition probably comes from **Aida-X**, **IK Multimedia**'s **Tonex**, **Amplifire**'s **Tonocracy**, **Native Instruments' Guitar Rig Pro 7** and **Neural Amp Modeller**.

$ $79.99. (But free to those who held a Two notes account prior to the release of Genome.)
W www.two-notes.com

MATT HOUGHTON

Gainlab Audio GLA-MP1 Bishop

Valve Channel Strip

Gainlab's channel strip delivers high-quality valve preamplification and optical compression, along with some interesting tone-shaping options.

We've looked at a couple of Gainlab's products in the pages of *SOS* already. The Dictator variable-mu compressor (December 2021: https://sosm.ag/gainlab-audio-dictator) and the Empress equaliser (April 2022: https://sosm.ag/gainlab-audio-empress) both impressed us with their combination of good build quality, great sound and value for money. So when I was offered the opportunity to test their Bishop Channel, a recording channel strip, I was keen — and I wasn't disappointed.

Overview

The first stage in the signal path is a valve mic/line/instrument preamplifier. Its large, stepped gain knob is surrounded by a busy-looking legend, with two concentric circles; each one indicates the setting for a different gain mode, with the modes selectable using an adjacent metal toggle switch. In the low-gain mode, this stage can apply up to 40dB boost, while in high-gain mode it can add up to 60dB. All this is achieved using three vacuum tubes. A neat yellow lighting system clearly indicates which mode you're in, with two backlit circles around the legends as well as the switch LED lighting up to give you that confirmation, meaning you don't have to squint to see how the toggle is set: a pleasing combination of style and substance!

There are seven metal toggle switches nearby, most relating to the preamp. Three are for features you'd expect on any modern preamp: 48V phantom power, a pad (for the mic input only) and polarity inversion. There's also a power on-off switch for the whole unit, a selector switch for mic, line or instrument mode, and another to choose between a front-panel input and one on the rear. A dedicated saturation stage for the instrument input allows you to add some colour to clean DI signals using a smaller Drive knob that's placed top left of the main gain control. As well as allowing deliberate coloration of the sound, this can be really helpful when used more subtly to tame any spikiness in a guitar signal before it hits the valve stages. I was a little surprised to find on some online images that the front-panel input appeared to be an XLR, rather than a combi type into which you could plug a bass or electric guitar directly... but fear not! That must be an early version, since the review unit featured just such a combi. (It's perhaps worth noting that, when active, phantom power is present on the XLR whether you have the mic, line or instrument setting selected.)

To the right of the gain knob and in the centre of the 2U rackmountable unit is a large backlit moving-coil VU meter that displays the output level. A recessed preset allows you to calibrate the meter's zero point with a small

screwdriver or similar, and next up we have the compressor stage, with its own LED metering. This is an optical compressor that, I think, is broadly based on the Teletronix LA-2A design, but it's not a direct clone and is actually more sophisticated. While it's a single compressor (there's only one optical cell) there are, as Gainlab put it, two slopes at play. Each has its own threshold control and independent 11-LED gain reduction meter. The top 10 LEDs indicate 1dB of gain reduction each, so the meter has plenty of resolution in the most useful range (if you're crushing a signal by more than 10dB, who cares what the meter says?). Beneath the Threshold controls are chickenhead knobs for Character (a knee control, I think) and Time Constant. As on most compressors of this type, there aren't separate attack and release controls but there are five switched attack-plus-release settings, ranging from Slowest to Fastest, plus a sixth, programme-dependent position.

The last knob, an output attenuator, has Off, -3, -6, -8 -10 and -12 dB settings, so you can bring down the output while increasing the gain or backing off the compression, allowing you to maintain sensible signal levels and judge changes in tonality without loudness really coming into play. (There certainly *can* be changes in tonality if you drive the input stage, even before you bring the compressor into the equation.)

Above this knob is a toggle switch to (dis)engage a high-pass filter, with Off in the centre position, a 60Hz filter below and a 120Hz filter above. So there's scope to tackle unwanted LF rumbles on their own, or to counter any unwanted proximity-effect bass lift. Finally, we come to the Bishop's secret weapon: another tiny toggle at the top right that engages a dynamic air-band boost at either 20 or 30 kHz.

Around the back, there are separate XLR inputs for mic and line sources, meaning you could for instance have the mic input hooked up to a wallbox in the live room, the line input hooked up to your patchbay, and still have the front-panel input available for mics and instruments. The front- and rear-panel mic sockets are wired in parallel but switched automatically, with the front-panel XLR taking precedence when a cable is plugged into it (plugging an instrument into the centre jack won't break the connection to the rear mic XLR, though). Also on the rear are an output XLR — Gainlab quote an output impedance of 150Ω — a ground-lift toggle switch, and an IEC mains power inlet with built-in switch and fusebox.

In Use

So, that's the quick tour completed, but the all-important questions are how the Bishop sounds and how easy it is to coax the desired sound out of it — and the headline response is that it sounds superb and it's easy to use!

The valve preamp has plenty of range, and when applying the amounts of gain you'd typically want in a studio session for normal mic and DI applications, there's impressively little by way of noise and distortion. The frequency response seems pretty much flat, too, but there's definitely character available here if you crank the gain and pull down the »

Gainlab Audio
Bishop Channel
$2450

PROS
- Lovely valve preamp that can handle mic/line/instrument signals.
- Combi input on the front.
- Versatile gain.
- Intuitive dual-slope compressor.

CONS
- A low-pass filter could be handy.

SUMMARY
A gorgeous-sounding combination of valve preamp and optical compressor, with enough bells and whistles to make it a versatile choice in the modern studio.

The Bishop offers separate inputs on the rear for mic and line sources, which will make integrating it into a patchbay trivially easy.

» output. It's a character that, as with all the best tube gear, isn't what I'd call 'warm' exactly (one man's warm is another man's 'mush'!) but more in the 'lively' and 'classy' ball park, with a slight rounding of transients and a little pleasing 'flair' to the high end. It's not dissimilar to the sort of thing I've heard when testing, say, Thermionic Culture or Manley gear, and not something that's ever likely to disappoint. This is a characterisation that applies especially to vocals, which always came out sounding wonderful.

The instrument input isn't unpleasant either. The discrete Drive circuit can add some really pleasing colour, although it's worth noting that for instruments with passive pickups that happen to need bags of gain, I found it hard to get a truly clean sound without really cranking the tube Gain stage — and that, at the sort of levels I sometimes needed, this meant there a fair bit of hiss was audible. It's not a problem as such, as it's very easy to roll off the top end of a DI'd guitar or bass part before feeding the signal to, say, an amp simulator. Nonetheless, it would have been nice to either have more clean gain available on the instrument input, or perhaps a switchable low-pass filter (maybe as another setting on the Dyn Air knob) to counter such boosted hiss.

The compressor has that forgiving personality that should be familiar to anyone who's used something like an LA-2A or LA-3A: somewhat relaxed and very easy to set up. Even though those dual slope controls might look complicated, they're actually intuitive enough that you can find your way around and set up a pleasing sound without thinking too hard about it; it's really just a case of setting the threshold for the desired gain reduction and then twisting the second slope knob to see what more it brings to the party — and it's a useful way to shape the sound. You can use this compressor for graceful control of vocals, guitars and basses, or anything else you might usually use an opto compressor for. If you wish, you can really slam your drum room mic too, and I really enjoyed doing this on a parallel setup, where I could use the valve stages to add a bit of grit before clamping down on the transients and bringing up the drum sustains. Of course, you'd need two units for stereo... Similarly, I loved using the Bishop as a parallel distortion effect on vocals, though note that, as there's no wet/dry blend on board, you'd have to set that up separately Still, while this might have been a nice addition, I usually prefer to EQ the parallel path anyway.

The air band and high-pass filter are worth a few words, since the numbers on the faceplate don't tell the whole story. For example, the HPF's 60Hz setting rolls off fairly gently, starting in the low hundreds and down by about 7.5dB by the time you get to 100Hz. So it's definitely an audible effect, not just a rumble filter. The same goes for the 120Hz setting, which rolls down from about unity at 2kHz to -10 around 250Hz. Similarly, the air band's frequencies might seem very high on paper, but the curves reach well into the audible spectrum. It's nice that this band is dynamic, too, since it avoids raising things like tape hiss and the noise floor of analogue electronics when there's no signal present.

Signal Flow

Having said in the main text that things are intuitive — for the most part, they are — some might find it confusing that the left-to-right channel layout doesn't mirror the signal flow. If you like to think your way through your processing decisions in that way, then, it's worth knowing that the input signal hits the compressor before it hits the main gain stages (which explains why the compressor's LED meters can be seen dancing busily even if the Gain knob is set at its minimum position). The line (18kΩ) and instrument (500kΩ) inputs are buffered, while mic (2kΩ) signals are brought up to level courtesy of an input transformer (the pad comes before this). The signal then flows through the input Gain stage, the high-pass filter, a +20dB gain stage (courtesy of an E88CC valve, this is engaged by the high/low gain switch), before finally hitting the dynamic Air band. At the end of this chain, the signal is fed to another E88CC which feeds the internal side-chain and the main output transformer. The signal then flows to the Phase (polarity inverter) and output attenuation stages, after which the signal is delivered to the XLR output and paralleled to the output level meter. (The specified maximum output level is a healthy +30dBu.)

Verdict

There's plenty of competition when it comes to valve optical compressors, but this isn't the most expensive option out there, and I reckon it's better than anything else I've used at its price. The sound is everything I'd want of a tube preamp and, with the dynamic air band, more besides. With mic, line and instrument inputs, as well as front-panel access for mics and guitars, it could work nicely in any studio, whether a musician/songwriter's small home-studio setup or a pro studio's rack. It's packed with more facilities than its closest competition, and while that does makes the striking front panel appear a touch busier than most, it's far easier to use than it might appear. In short: highly recommended!

$ $2450
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Present Day Production MUM-8

Active Monitors

Can a direct sales model and high-tech components make top-level monitoring affordable? Present Day Production say so.

PHIL WARD

People get into designing speakers for all sorts of reasons. Sometimes, there's a family connection. Sometimes it starts off as a hobby or a teenage obsession (as in my case). Sometimes a choice made in education leads directly to a career. The subject of this review, however, doesn't really fit any of those narratives. The MUM-8 active monitor is the brainchild of Mark Ashfield and James Nugent who, along with running a successful mix and mastering business, also create and present the Present Day Production YouTube pro-audio review channel.

Mark and James' decision to design and manufacture a monitor was driven partly by a desire to install Atmos monitoring in their studio, and the realisation that this wouldn't be affordable using their preferred ATC monitors. So they adpoted the radical solution of building their own monitor, and after a year or two spent learning and developing, the MUM-8 was introduced to an unsuspecting world. A further respect in which Present Day Production are unusual is that rather than doing business via dealers or distributors, they sell direct to end users. This business model offers a significant advantage in terms of costs, but it also potentially leaves distant end users without local support. Present Day Production say they overcome that issue by offering online help that can even extend to the remote configuration of monitor EQ.

MUM's The Word

The MUM-8 is a three-way closed-box active monitor of dimensions that put it, to my way of thinking, somewhere between a large nearfield and a compact midfield. At 19.25kg the MUM-8 isn't light, but not so heavy that weight is likely to become a significant installation issue. The three drivers comprise a nominally 200mm (8-inch) aluminium-diaphragm bass driver, a 75mm (3-inch) silk-dome midrange driver and a 25mm (1-inch) silk-dome tweeter. The drivers are all bought-in OEM devices; from Purifi of Denmark in the case of the bass driver and Bliesma of Germany in the case of the midrange driver and tweeter, and they're all impressive examples of the driver designer's art.

Rather than being the affectionate yet respectful term for a female parent, MUM stands for Modular Upgradeable Monitor. The '8', if you hadn't already guessed, refers to the bass driver diameter. The MUM-8 is modular in that its rear-panel-mounted amplification module can be detached and connected by speaker and control cables to enable, for example, the monitor itself to be soffit-mounted. And the MUM-8 is upgradeable in respect of its midrange and tweeter drivers. The standard-issue silk-diaphragm drivers can be swapped out for beryllium-diaphragm units. Mark and James say that doing so brings a significant performance upgrade, but for review purposes it seemed to make most sense to look at the standard version.

The MUM-8 occupies a conventional format, with a tall and narrow cabinet that has drivers stacked asymmetrically one above the other on the front panel. However, in some nearfield installations the stacked driver arrangement is likely to leave the tweeter located somewhat above a seated nearfield listening position. So the MUM-8 provides the facility to swap its tweeters, with a similarly dimensioned circular blanking plate located to the side of the midrange driver to enable the monitors to be installed in landscape orientation yet still have the midrange driver and tweeter stacked vertically. This means landscape orientation doesn't suffer from the potentially compromised horizontal dispersion in the mid/tweeter crossover band of laterally arrayed drivers. It's a simple solution to a very real problem. Most of my listening to the MUM-8 was done with tweeters arranged for landscape orientation.

Rigid Thinking

In expressing an almost entirely functional aesthetic, the MUM-8 won't win prizes for looks. Its cabinet is extremely inert, and in monitors built primarily to do a professional job, panel rigidity probably trumps beauty. The Valchromat from which it's made inhabits the same kind of material world

as MDF but offers a few advantages: it's stiffer than MDF, can be machined more effectively, and is coloured all the way through (its base fibres are dyed before being pressed into panel form). It can therefore be used without need for any final finishing, which is how it's employed in the the MUM-8.

The MUM-8 rear-panel connection facilities, along with mains power, comprise balanced and unbalanced analogue, and S/PDIF, optical and AES digital inputs. Input selection is carried out using a compact remote control handset supplied with the monitors. Channel selection for digital inputs is preconfigured so that installing a pair of monitors with their midrange and tweeter drivers positioned towards the inside results in the correct left/right channel routing. This principle also holds for landscape orientation.

With three Hypex Class-D modules rated at 250 Watts each for the bass and midrange drivers and 100 Watts for the tweeter, the MUM-8 is not short of power. Along with power amplification, the MUM-8 Hypex module also incorporates a DSP element that provides analogue-to-digital conversion (at 24-bit/192kHz), crossover filtering, three preset EQ profiles and, potentially, the option for user-defined global EQ to be configured and applied.

As I mentioned in an earlier paragraph, the MUM-8 amplification module can be detached from the back of the monitors and located remotely. When attached to the monitor, the module is connected to the drivers and front-panel display using short Speakon-terminated speaker and multi-pin XLR-style-terminated comms cables; when it is to be located remotely, these need to be swapped out for extended cables. The comms cable drives a front-panel OLED display that shows input gain setting, input selection and EQ preset. Three EQ presets are available: P1 is the default, wide-bandwidth (extended bass) option. P2 offers less low-frequency bandwidth but consequently higher maximum volume. And P3 offers a low-frequency roll-off suitable for integration with a subwoofer. Presets are selected, as are inputs, using the remote-control handset. The MUM-8 DSP also enables wide-band EQ to be configured and applied to the monitor to suit personal preferences, and even perhaps to provide some rudimentary room optimisation. However, this needs to be done from a PC running a proprietary Hypex DSP configuration app connected to a service USB socket on the amplifier module, so is perhaps not something for the inexperienced.

Driving Forces

I promised a little more detail on the drivers, and I'll start with the bass driver. Its unusual roll-surround is the result of a complete reappraisal undertaken by Purifi of the sources of distortion inherent in the traditional moving-coil driver architecture. One of those distortion sources, which has gone unnoticed for decades, is that conventional roll-surrounds tend to behave asymmetrically. Put simply, the radiating area of the surround is different depending on the direction of diaphragm movement, and that leads to distortion. So the surround of the MUM-8 bass driver is designed not only to do the job of terminating the diaphragm while freely allowing axial movement over large displacements, but to do so without behaving asymmetrically. It's the combination of those two requirements that results in the surround's highly unusual and complex shape. While the surround is the most visible result of Purifi's analysis of driver distortion mechanisms, there are numerous other innovations hidden within the driver's motor system (its magnet and voice coil). Its pole-piece, for example, rather than being a simple steel component, is also a secondary flux-control magnet, and the driver voice-coil is wound variably, rather than linearly, in order to compensate for non-linearities that occur with diaphragm movement.

One of the benefits of the Purifi approach is that their drivers are able to retain linearity over extreme diaphragm displacement. The driver employed in the MUM-8, for example, is generally linear to ±9.5mm displacement — almost twice as much as conventional drivers. This means it's possible to engineer extended low-frequency bandwidth in an active monitor without having to employ reflex loading, with its inherent time-domain and compression issues. As a result, the MUM-8 low-frequency cutoff specification is -3dB at 26Hz, which is pretty much subwoofer territory. There's currently little else around quite like Purifi's driver technology, and its use in the MUM-8 constitutes a notable differentiator (the only other pro monitor company currently employing Purifi drivers that I'm aware of are Wayne Jones Audio).

The midrange driver and tweeter are sourced from Bliesma in Germany. Like Purifi, the Bliesma company are relative newcomers to the high-performance driver business, but founder Stanislav Malikov has a long and distinguished career in driver design and manufacture with companies such as Morel and Accuton. The Bliesma product range comprises just dome tweeter and midrange drivers, offered in a range of sizes with a variety of diaphragm materials: from silk at the entry level via magnesium and diamond to »

The MUM-8's amplifier module can be removed and mounted independently of the speakers if desired.

Present Day Production MUM-8

£4995

PROS

- Incredible bass.
- Fantastically revealing.
- Brilliant stereo imaging.
- Modular and upgradable.

CONS

- Tonal balance needs tweaking.

SUMMARY

The MUM-8 displays a few rough edges but its fundamental quality and ability are remarkable. It's quite an achievement for a first effort.

beryllium. Bliesma drivers, like those from Purifi, are meticulously engineered within their motor systems to minimise distortion and any non-linear behaviour.

On The Bench

I took a MUM-8 to my large measuring space, fired up FuzzMeasure, and captured some acoustic data to investigate its electro-acoustic performance. The data was taken with the measuring mic 1.5m distant on an axis aligned midway between the MUM-8 tweeter and midrange driver. The nominal sound pressure level was 90dB at 1m: not particularly loud, but loud enough to have the MUM-8 flexing its muscles. Volume level is significant when measuring speakers, because distortion mechanisms become significantly more apparent, and potentially more audible, as volume increases.

Diagram 1 shows the axial frequency response along with its second- and third-harmonic distortion, and there's a few things to consider. Firstly, the axial response isn't the flattest I've ever measured. It displays gentle dips centred on 200Hz and 750Hz, and then between 2kHz and 7kHz there's some general untidiness. It's hard to know for sure what is responsible for these features, but the dip at 750Hz could well be a crossover artefact (the crossover between bass and mid drivers is nominally at 500Hz). Online reviews of the midrange driver suggest that the dip at 6kHz might be inherent to it, rather than to the MUM-8's design. However, with the crossover frequency specified at 3.5kHz, I'd expect a 6kHz midrange driver dip to be less apparent — although that does depend on the crossover filter roll-off rates used.

A 'flat' response might seem to be one of the prime requirements of a loudspeaker, especially a monitor. However, even leaving aside the extremely un-flat frequency response we're likely to perceive thanks to the influence of the listening environment, the idea of a flat frequency response at source (the speaker) becomes increasingly nebulous the more you examine it. For example, the MUM-8 axial frequency response I measured would have looked somewhat different if I'd chosen a different microphone position or distance (see Diagram 2). Similarly, I've chosen to display the response with a bit of curve smoothing applied to make it a little easier to comprehend. I chose 1/48th-octave smoothing (not much), but if I'd gone with 1/6th-octave smoothing (which psychoacoustics tells us is more representative of how we hear), the curve would look different again. So, while frequency response measurements give us a handle on the baseline characteristics of a speaker, and measuring one that's really flat always gives me a buzz (get a life, Phil), measuring a frequency response that raises questions doesn't necessarily mean the speaker is fatally flawed. Far from it.

> **"The bass is perhaps even better than I hoped it would be, combining fabulous bandwidth with outrageously seductive speed and detail."**

And if you wanted an example of why that's the case, the green and blue curves in Diagram 1 provide it. These respectively show second- and third-harmonic distortion, and both are low, but the more subjectively problematic third harmonic is really very low — generally better than 60dB or more down above 100Hz. -60dB is equivalent to 0.1%, which is more like an amplifier distortion specification than it is typical of a loudspeaker, and I'd swap a few undulations in the frequency response for such low distortion any day of the week.

Ups And Downs

Diagram 2 shows the variation in frequency response 20 degrees vertically above (blue) and below (green) axis. The red curve is a repeat of the axial response. Moving the measuring microphone to an off-axis location vertically introduces relative time delays between the drivers, because the path lengths from each driver to the mic change. This means that audio from, for example, the midrange driver and tweeter that were in phase on-axis are slightly out of phase off-axis, and that's revealed in the extra level of untidiness in the off-axis frequency response curves. The dip around 750Hz in the -20 degree (green) off-axis curve is also perhaps due to the relative phase of the bass and midrange drivers changing, although the naturally increasing directivity of the bass driver as frequency rises may also be a factor. Overall, the off-axis response curves are generally typical of a speaker with spaced drivers — although, as with the axial curves, they are a little untidy. If I'd made the measurements with the alternative (landscape) driver arrangement in place, the results would be slightly different.

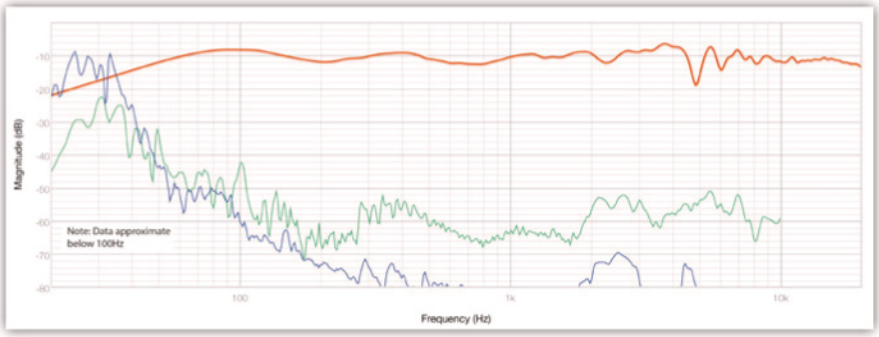

Diagram 1. MUM-8 axial frequency response (red), and second/third harmonic distortion (green and blue, respectively). All curves 1/48th octave smoothed.

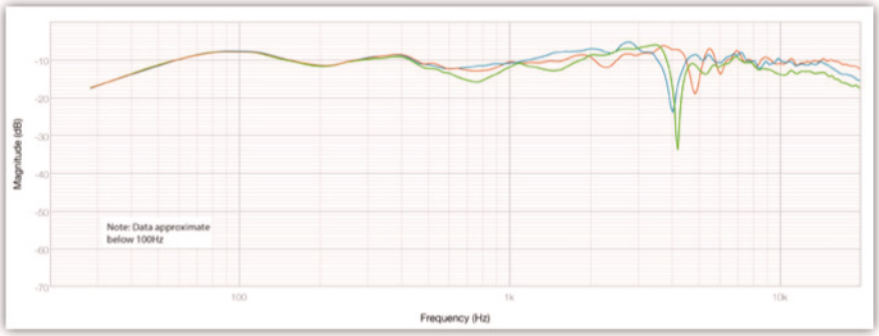

Diagram 2. MUM-8 on-axis frequency response (red), with +20° (blue) and -20° (green) vertically off axis. All curves 1/48th octave smoothed.

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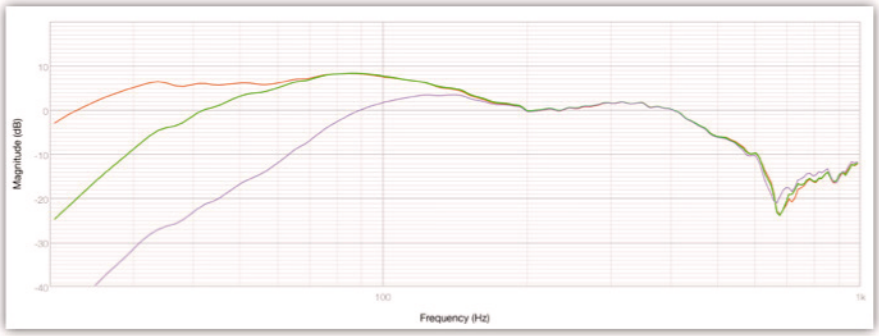

Diagram 3. MUM-8 close mic LF frequency response. Red - Preset 1, Green - Preset 2, Purple - Preset 3. All curves 1/48th octave smoothed.

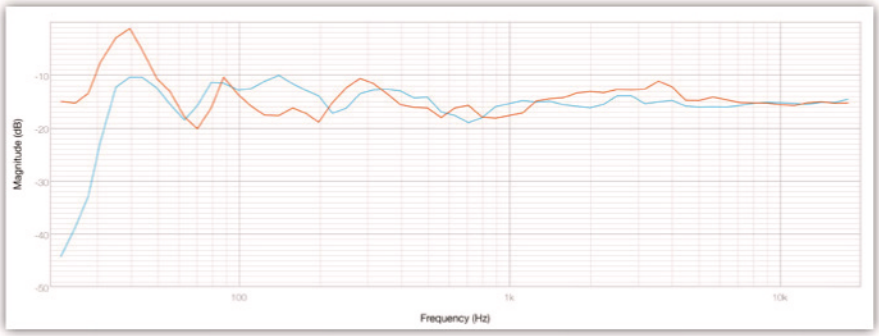

Diagram 4. MUM-8 (red), and Neumann KH150 (blue, room optimised) in-room response. All curves 1/6th octave smoothed.

Diagram 3 illustrates the low-frequency bandwidth characteristics of the MUM-8. The data was captured by placing the measuring mic less than half a centimetre from the bass driver. The three curves illustrate not only the overall bandwidth achieved by the MUM-8 but also its preset options. The red curve shows Preset 1, the green curve Preset 2, and the purple curve Preset 3. The first spot is that the 26Hz/-3dB low frequency cutoff specified by Preset Day Production looks reasonably genuine, although this does depend a little on where you choose to place 0dB. The other preset options result in -3dB at around 55Hz and 90Hz — the latter, I think, a sensible choice for integration with a subwoofer. A slightly strange feature of Diagram 3 is that the bass driver appears to show a gentle peak in response around 100Hz followed by a down-shelved region between 200Hz and 400Hz, above which the low to mid crossover filter begins to become apparent. The down shelf above 100Hz appears to tie in with the 200Hz dip revealed in the axial frequency response in Diagram 1.

Into The Detail

Monitors are of course primarily for listening rather than measuring, and I began my tests with the MUM-8s set up in landscape orientation on my monitor shelf. I had really high hopes for the bass. With one of the world's highest-performance bass drivers, in a closed-box enclosure and with generous amplifier power available, what could possibly go wrong? As it turns out, nothing. The bass is perhaps even better than I hoped it would be, combining fabulous bandwidth with outrageously seductive speed and detail. But it was the explicit communication of pitch that really caught my attention. The sometimes hazy reproduction of, for example, double-bass pitch and kick-drum tuning that we kind of accept as inevitable in so many monitors, small ones especially, is completely absent with the MUM-8. No double-bass player will ever again get away with dodgy 'E' string intonation because the monitors can't resolve it. This element of the MUM-8 is all but worth the ticket price alone (and I say that as a double-bass player who sometimes struggles to stay in tune).

The MUM-8 is more than a one-trick donkey, though. Further up the band, the quality of the Bliesma midrange driver and tweeter are instantly apparent with their immense clarity, inaudible noise floor and uncannily sharp stereo image focus. The MUM-8 is an incredibly revealing and analytical monitor that immediately lays all the elements of a mix naked and exposed. You almost feel you ought to be able to reach into the stereo image and touch things, and the level of detail occasionally borders on the unsettling. I do have a gripe, though, which is that I found the overall tonal character a little unbalanced. The 200Hz dip revealed in the measurements is subjectively noticeable, leaving male voices and lower-register acoustic instruments lacking in body and warmth. Cellos sounded somewhat viola-like, for example. At the same time, the upper presence band (say, 2kHz to 4kHz) is, I think, a little over-emphasised, as is the tweeter band generally. The MUM-8 takes no prisoners tonally, which could result in mix translation challenges and can make it a slightly tiring listen. Some in-room response measurements I made illustrated in Diagram 4 confirmed both the 200Hz dip and a couple of dB too much upper midrange compared to my reference (and room optimised) Neumann KH150s.

My tonal balance doubts about the MUM-8 might appear to be a significant flaw, but I refer you to an earlier paragraph where I wrote that the MUM-8 has some significant DSP power embedded within its electronics that includes comprehensive EQ facilities. Present Day Production also say that they are perfectly happy to work with customers to tweak the EQ to suit needs and preferences. So my tonal balance doubts could be easily fixed, and I proved that to my own satisfaction simply by applying some EQ from a DAW plug-in; +2.5dB at 200Hz and -2.0dB from between 2kHz and 6kHz did the trick for me. It left me with a more comfortable tonal balance but cost none of the MUM-8's fundamental quality. And it's that phrase, "fundamental quality", that sums up the MUM-8 for me. Take three of the best drivers available, put them in a well engineered, rigid enclosure, drive them with powerful and high-quality amplification, and you get the MUM-8. I've just made it sound really simple, but in reality it takes huge commitment, hard work and not a little skill. Mark and James ought to feel very proud of what they've achieved. ■■■

ALTERNATIVES

Despite its direct sales model, a pair of MUM-8s is still a high-end monitoring option. Among many others, alternatives worth considering in a similar price bracket might be the **Focal Trio11 Be**, **PMC 6**, **Genelec 8341A**, **PSI A21M**, **Mesanovic CDM65** and **ADAM S3V**.

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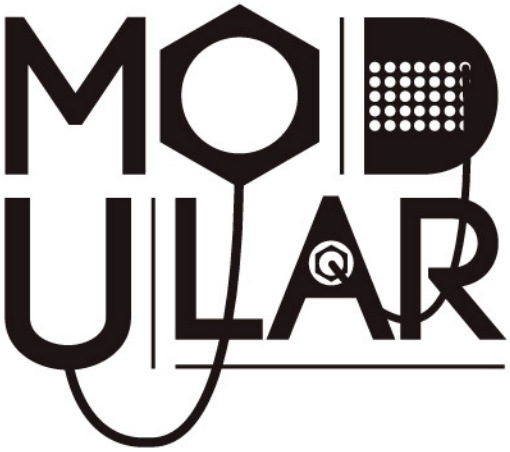


Bastl Instruments Neo Trinity

Eurorack Module

I must confess that when the Neo Trinity arrived, I was so inspired by its looks that I put it back in the box and left it on a shelf. Little did I know that behind the squashed text and multi-level shift-button action lies a throbbing agent of change. It's a patch catalyst, a virus-like modulation engine that can infect your rack with movement, rhythm and automation. It's ace!

The Neo Trinity has evolved from Bastl Instruments' CV Trinity, an unappreciated six-channel modulation generator that has much of the functionality but none of the style of this new version. Like many recent modules, such as the Pizza, Basil and Ikarie, the Neo Trinity is dark, brooding and overflowing with possibility. It brings the talents of the CV Trinity to a smartly ordered and complex form. Complexity will be a nagging feature.

Put simply, each of the six channels can be an LFO, an envelope or a loop of recorded CV. With some minor shenanigans, these three basic modes can turn the Neo Trinity into a funky little groovebox. Six LFOs could prove very useful, but alternatively, you could quantise movement into notes, tap envelopes into your filter or VCA, and pull up trigger patterns to run your drums. The versatility here is brilliant, and everything has recordable/automatable elements that turn it into quite a party.

There are lots of green LEDs. Some give you shapes, others give you pulses and triggers, while more show you what combination of things is active for the given combination of button pushes. There's lots to touch, lots to fiddle with and lots of indications to show you when you have done so. When patched up, it looks busy, tight and uncomfortable, like wearing a suit that's two sizes too small, but somehow, the Neo Trinity has enough elastic in its waistband to keep it all from spilling over.

All the action revolves around the two big buttons right at the bottom and the very fingerable knob. You hold down the Rec button and either move the knob or hit the Shift/Trig button to record movement or triggers into that channel. In LFO mode, you can record rate changes on the knob and reset the waveform on the trigger. In Envelope mode, you can tap in a series of triggers, and then the knob controls the decay. In CV mode, the knob creates your modulation pathway. All of this can be generated on the fly, cycling around your clock, building up channel after channel of looping modular manipulations.

With a few other button holds and selections you'll find that the CV recording can be quantised to notes for writing quick, knob-turned bass lines and melodies. Holding some other buttons in envelope mode puts in preset patterns for instant percussive tracks. And then there's a mute button for dropping channels in and out. A double-tap erases the triggers or CV in an instant, ready to try it all again. Channels E and F have CV inputs that can scale and affect the outcomes, and a Meta CV input can do the same to any selected channels.

Although the Neo Trinity feels overstuffed, once I got through the blindness of interacting with black buttons on a black background and acquired some muscle memory on the combos, it was tremendous fun to use. Just the act of being able to tap in a rhythm for some envelopes with my filter brought a massive smile to my face. Once patched in, I would play and play, sometimes getting lost in the wrong shortcuts, but more often than not, I was lost having a thoroughly good time.

Robin Vincent

$ $320

W www.bastl-instruments.com

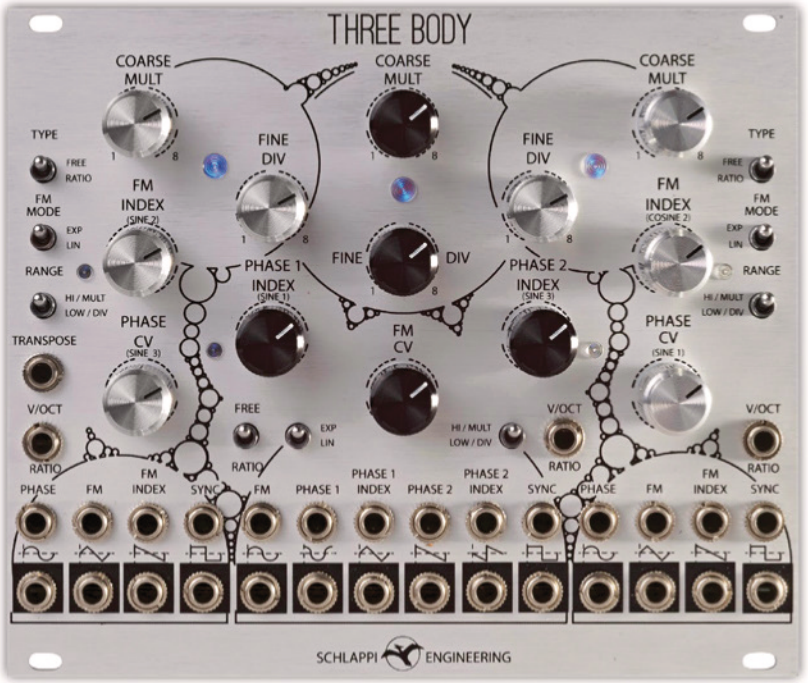
Schlappi Engineering Three Body

Eurorack Module

A "phase and frequency modulation toolkit" is how Eric Schlappi describes the Three Body. Schlappi himself is an electronic musician dealing primarily in the world of noise (I myself am a proud owner of two of his cassettes), with his modules seemingly designed to facilitate his own craft as much as they are for others. A digital module with all the upfront goodness of an analogue interface, at its most basic level the Three Body is an array of three independent oscillators which can each output sine, triangle, saw and square waves, with the central, main oscillator adding 90-degree phase-shifted cosine and co-saw waves on top. That's a total of 14 separate outputs, and it has the same number of inputs, too.

Alongside conventional volt-per-octave CV inputs for each oscillator, there's also a Transpose input that handles all three at once. Minimoog-style triple-oscillator massiveness, triad chords — it all sounds excellent and is incredibly quick to dial in. Atop this, the central oscillator's out-of-phase waveforms are primed for lush stereo width. If the Three Body was sold as a classic oscillator then I dare say it would already do its job well. Things start to take off, however, with the involvement of its various switches and modes.

The Three Body behaves most like a conventional oscillator module when the three oscillators are in Free mode, where they operate independently of one another, and with their controls for FM Index, Phase CV and Phase Index on zero. Switch any or all into Ratio mode and move any of these knobs even modestly, and you will be catapulted into another sonic territory entirely. This is largely thanks to the fact that the Three Body operates in the realms of both phase and frequency modulation, which, for brevity, I'll just say are closely related but not the same. The oscillators will harmonically track another signal patched to the sync input according to the ratio setting of the Mult and Div knobs (whose functions



in Free mode constitute Coarse and Fine tuning respectively). An FM Mode switch toggles between linear and exponential settings, so it's possible to track an incoming pitch accurately or, well, not so accurately. Last up is a Range switch which, ostensibly, influences the tuning range in Free mode and switches between multiplication or division of frequencies in Ratio mode.

I say 'ostensibly,' because there is simply a vast amount going on under the hood here, and I must resist the temptation to simply list combinations. You'd be hard pressed to fill a 25-page manual with information about most standalone Eurorack oscillators, but here it's as if Schlappi could easy have written a whole book about the Three Body. The manual usefully includes a generous amount of theoretical explanation of the difference between phase and frequency modulation, and a reference table for looking up all the possible switch combinations and what they each achieve. There are eight combinations, by the way, but that's just for each oscillator, remember.

It's ripe for internal patching, and there are some normalled connections as well; the sine wave of oscillator 3 is normalled to the Phase CV of oscillator 1, for instance, and the sine wave of oscillator 1 to the Phase CV of oscillator 3. Oscillator 2 has its two Phase Index inputs normalled to receive the sine waves of both its neighbours. So, editing anything on any one oscillator can have a potentially huge effect on the whole equation. It's essentially the oscillator equivalent to holding two mirrors up to one another; I'm sure there's an equation out there to calculate the total range of this thing, but rest assured you are unlikely to find its limit.

To best understand the Three Body from a basic level, I found that a good starting point was to dummy-patch all the CV inputs, just to get some idea of what the hell was going on. It's easy to forget that those normalled connections are just one means of operation. Replace even one with, well, anything, and the Three Body reinvents itself all over again. It's as if every single move simply opens up more options.

It's sometimes a good idea to disregard everything you've learned about synthesis, throw caution to the wind and start turning knobs with wild abandon. In many ways, you may as well throw all the theory out the window (or gratefully leave all the theory to Eric Schlappi) and just turn the Three Body on and start playing. I guarantee you'll find a sound you like. Sure, manic knob-twisting means that, at worst, things can descend into harsh and squelching generic FM noise, but in the spaces between there's something exceptional here. At 30HP it's the largest module in the Schlappi range by some way. But considering what's possible with it, the Three Body feels positively pint-sized. Schlappi Engineering might just have made a modern classic.

William Stokes

$ $595

W www.schlappiengineering.com

What's New

Winterbloom have recently unveiled the Neptune, a resonant diode ladder filter designed in collaboration with Carson Walls of **Decapod Devices**, which delivers what they call "delightfully strange behaviour" with a feedback-modulated distortion circuit named 'Salt'. www.winterbloom.com

Expert Sleepers have released another addition to their line of all-analogue modules: a compact and flexible multi-LFO module named Otterly. www.expert-sleepers.co.uk

On the other side of the world, meanwhile, brand new Australian developers **Cycle Instruments** have unveiled their inaugural module. Tetrachords is a four-track polyphonic CV and MIDI note sequencer that promises brains, beauty and brawn with opulent amounts of connectivity and control, alongside a novel scale-based system for generating notes. www.cycleinstruments.com

Finally, German developers **ST Modular** have taken their Pedal Pal effects pedal interface module into stereo land with the Pedal Pal Stereo, a rather gorgeous effects loop with expression CV control, tone control and even a phase alignment button. www.st-modular.de

1010music Bluebox

Eurorack Module

When it arrived back in 2020, the desktop version of 1010's Bluebox mixer and recorder was in many ways a game-changer. Without a slider or pot in sight, its astonishingly simple interface comprised little more than a screen, four encoders and a dozen or so buttons, arranged on a panel of five by five inches. On top of this, it could record and play back hours of audio, and as 48kHz, 24-bit WAV files, no less.

I write all of that in the past tense, but of course the desktop Bluebox is still very much available — and a touch cheaper than the Bluebox Eurorack Edition, which at its core is the same beast. Up to 12 mono tracks of audio (or six stereo tracks) are on offer, each with their own gain, volume, pan, four-band EQ, on-board reverb and delay. As if a compact mixer could offer any more, it also has a decent-sounding and flexible

» bus compressor to help glue everything together at the other end.

The Bluebox's interface, centred around its touchscreen, is flanked by four main encoders and two navigation buttons, named A and B. Eight rubberised buttons sit below the screen: these are used to cycle between different modes, and there are also three larger buttons that operate its recording and playback transport. Since it doesn't have dedicated hardware controls per track, the Bluebox workflow, operating across both its touchscreen and hardware controls, revolves around choosing an appropriate screen for whatever function you want to focus on. Mixer mode allows a specific parameter, such as volume, to be viewed and adjusted across several tracks, while Track mode focuses on a chosen track, allowing all its different parameters to be viewed and adjusted in one place. It's not perfect, and it can feel a little menu-heavy, but it is sufficiently open-ended to allow the user to find their preferred way of interacting, and it didn't take long for me to find mine. Speaking of which, like its predecessor, the Bluebox is best mounted into a horizontal or angled part of your rack. I mounted mine into the lowermost vertical row of my system, which suited its input and output positions well, but soon found craning my neck to navigate its touchscreen at such an angle a little tiring.

Several extra jacks adapt the Bluebox Eurorack Edition into modular land, namely six CV inputs and an analogue clock output. Be warned, though, that it does require a lot of power, not to mention two headers on your busboard. The desktop Bluebox's six TRS inputs and two outputs are here exploded into 12 TS inputs and four outputs respectively, and while the inputs remain AC-coupled — meaning they cannot work with CV signals as well as audio — the outputs are DC-coupled, meaning that they can. That's a serious amount of I/O — surely as much or more than I've seen on any other Eurorack mixer, and with a headphone output on top to boot. The main takeaway here is that the Eurorack Bluebox can nestle right into your patch with effects and modulation, as well as handling its final output stage.

On top of this, with USB-C ports and upgraded capabilities compared to the desktop version, the Eurorack Bluebox can also send and receive up to two channels of audio to and from your computer and DAW, essentially becoming a fully functional audio interface. Audio over USB, I'm told, was added in response to popular request, so many will doubtless be very happy to see it here, although it would have been nice to have had more than two channels. A second USB port can be used for MIDI control, though not with a computer. Since there's capacity for both Type A and B TRS MIDI here, as well as its assignable CV inputs, it's fair to say you'll hardly be left wanting when it comes to remotely controlling channels' levels, effects and more.

Even if its effects didn't sound ace, and even if it couldn't integrate creatively with your patch, the Bluebox would still be a highly powerful module. It feels almost like a miniature DAW, crammed into just 30HP (although it hasn't really been miniaturised — this is more or less the exact size of the desktop version) and the capacity to record part or all of your patch at the drop of a hat is phenomenal. There are mixers, and there are mixers you can almost, well, mix on. Far from simply drilling holes in its faceplate, 1010music have given this edition a real *raison d'être* in Eurorack while still staying true to what made the original Bluebox such a success. Doubtless this version will enjoy similar recognition.
William Stokes

$ $699
W www.1010music.com

External Instruments

Modular Technique

The relationship between live external instruments and modular has always been a somewhat awkward one, but it's nonetheless a hugely worthwhile avenue to explore, and you may just open up a side of your system you never knew was there. Signals from microphones or DI'd electric guitars will need boosting to the appropriate level. Modules like the recently reviewed Cosmotronic Cosmix Pro mixer and Intellijel's Audio I/O can handily boost incoming audio by 18dB and 20dB respectively, for instance. For a little more character, I highly recommend Future Soundsystems' fantastic Throbbing Gristle collab, the TG5 Gristleizer preamp, which has a three-band EQ, variable gain with Boost switch and a Bite control for achieving tone comparable to many conventional guitar amps. Next it's a case of selecting (and even grouping) modules to process your audio. Boldly modulated filters and VCAs are a natural choice, while modules like the Nautilus delay, Aurora reverb and Mojave granuliser from Qu-Bit can catapult your signal into cosmic territory. Beyond audio, it's key to harvest a CV signal from your instrument: another string to the TG5's bow is that it also has a gate and trigger output, so strumming a guitar or yelling (ahem, singing) into a microphone can trigger envelopes, start and stop sequences and more. Play on!
William Stokes

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BEYONCE
BLAKE SHELTON
BRANDI CARLISLE
BRUNO MARS
CARRIE UNDERWOOD
COLDPLAY
DAVE MATTHEWS BAND
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EAGLES
ELTON JOHN
FALL OUT BOY
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GUNS N ROSES
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KORN
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MORGAN WALLEN
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PAUL MCCARTNEY
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Erica Synths Bullfrog Analogue Synthesizer

Erica Synths and Richie Hawtin's educationally inspired synth isn't just for kids.

ROBIN VINCENT

The concept behind Erica Synths' Bullfrog synthesizer is an unusual one. It comes from the noble idea that synthesis can be educational. The idea that, behind the bleeps, bloops, sweeps and fizzes, there's science, art and electronics that can, if given the opportunity, broaden the mind of the child. Everything about the Bullfrog synth is informed by the hope that it will find itself sitting on the graffitied desk of a Year 10 physics student and live to tell the tale.

Bullfrog comes from a collaboration between Erica Synths and electronic music producer Richie Hawtin, who was looking for a way to broaden access to synthesizers through schools. Erica Synths are no strangers to education. Last year, they released a series of educationally focused DIY kits that taught you the electronics behind the modules you were building. Erica Synths' lead visionary Girts Ozolins was once a high school maths and physics teacher. So, Bullfrog feels very personal, like it holds a lot of expectation and gravitas.

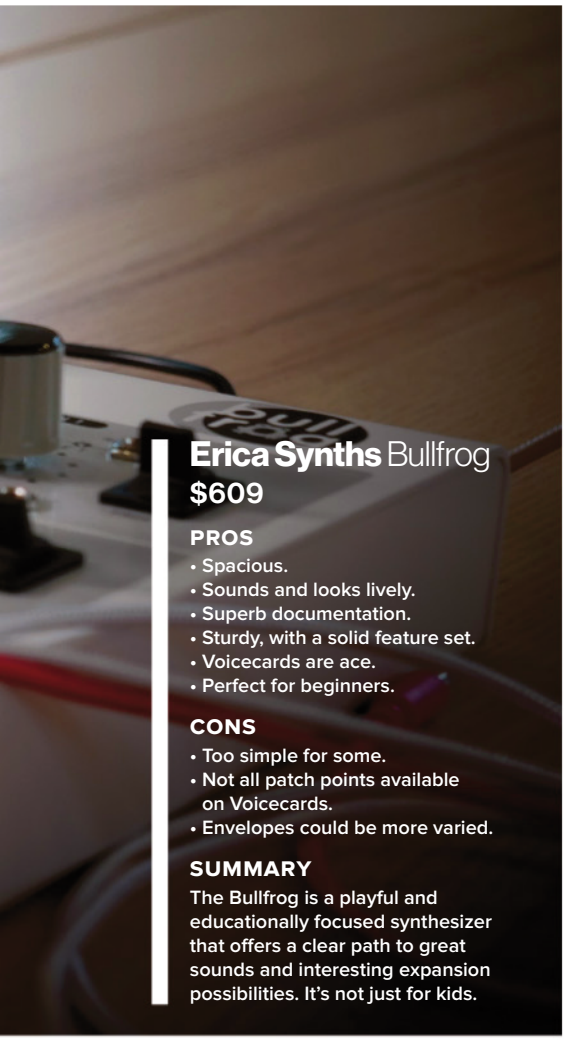
The question this raises for me as a reviewer is whether this is a science project for kids, or if Bullfrog can be a cool synth in its own right. Does meeting the needs of the educational environment make the product better?

Hardware

The first surprise is the sheer size of the synthesizer. It's a proper desktop machine, much bigger than the Eurorack device I was expecting. The modules could be described as being 4U in size, but the synth is spread out and more intertwined than on a regular modular system. It has a one-piece front panel, so while the signal routing may be modular, the reality of the individual modules is not. I wonder about the reasoning behind the modular/non-modular choice. On the one hand, the synth is screaming to be taken apart, but on the other, it needs to retain a consistent intention and form if you are going to teach it successfully.

The second surprise is the colour. If Erica Synths is anything, it's all about the black, and that Eastern European techno vibe. But here we have fun, we have light and joyfulness, we have strong child-like colours deliberately defining the elements within the synth. The Bullfrog probably wouldn't look out of place carrying diseases in the children's waiting area of your local GP surgery. It's a liveliness that may put off a lot of Erica's regular customers, but I feel totally drawn to it.

From left to right, we have a full-range analogue VCO in bold blue, a low-pass VCF in grateful green and a VCA with an unexpectedly integrated analogue delay



Erica Synths Bullfrog
$609

PROS
- Spacious.
- Sounds and looks lively.
- Superb documentation.
- Sturdy, with a solid feature set.
- Voicecards are ace.
- Perfect for beginners.

CONS
- Too simple for some.
- Not all patch points available on Voicecards.
- Envelopes could be more varied.

SUMMARY

The Bullfrog is a playful and educationally focused synthesizer that offers a clear path to great sounds and interesting expansion possibilities. It's not just for kids.

in rupture red. To the right we have two envelope generators, and below them a sample & hold circuit. Messing up the ordered lines under the VCO and VCF is a three-channel mixer in boastful black. At the top is a clock divider; although there's no clock as such, you can grab a pulse from the sample & hold circuit.

Across the top is an intriguing blue slot. We'll come on to that. This leaves us with a volume knob and an integrated speaker in the shape of a bullfrog. The speaker isn't fabulous, but it's loud enough and decent enough for classroom shenanigans. Thankfully there's also a mono output and a headphone output on the back, along with a MIDI In, CV and gate in and a USB socket for firmware updates.

The Bullfrog is big, chunky and dependably solid. It's great for both the sticky fingers of primary and the barely controlled grapples of secondary. While there is something undeniably toylike about it, this is a serious piece of hardware and the Bullfrog feels like it will survive the most challenging of environments.

Signal Flow & The Manual

The Bullfrog is modular, so if it is unpatched it won't make any sound. The front panel is very clearly labelled, and all the patch sockets have a direction-of-flow indicator that runs along a line to the thing they're controlling. However, it wouldn't be immediately clear to a beginner how you'd get something out of it. So, initial modular bafflement pushes you to open the thick and colourful manual.

The manual guides you in choosing between patching the sine or pulse wave outputs to the VCA, or you can take Mix Out from the mixer, which also includes a noise generator, to the same destination. You don't have to patch directly into the speaker as there are certain elements behind the scenes that make these connections. Otherwise, Bullfrog follows the traditional subtractive routing of VCO to VCF to VCA, with envelopes patched in to control them.

Educational Realities

I am married to a teacher, so I have a pretty clear idea about the challenges the educational system presents to well-intentioned outsiders with their educational novelties. For any off-piste ideas to fly, they have to be backed up with the right resources so that a teacher can pick them up and teach with them. If it is too complicated or requires them to generate a bunch of lesson plans and learning objectives, then, with the best will in the world, the teacher will ignore the shiny box and run a lesson they are already familiar with to make the best use of that most precious of teaching commodities: time. Perhaps it's hardly surprising given Girts Ozolin's previous life as a teacher, but Erica Synths appear to be nailing this aspect.

Getting sound from the four available sources takes up four pages of the manual, which then takes you into the idea of plugging in a keyboard via MIDI or CV. It's nicely illustrated and builds on each patch, taking you through the filter and the VCA before pulling in the envelopes and sample & hold circuit. There are even four patch examples for the clock divider.

I don't think I've ever seen such a great instructional guide to a synthesizer. It's clear, straightforward and pitched really well at around high-school age. I've already learned a couple of things I hadn't considered before. The first third of the book covers everything from the nature of sound, pitch and timbre, through to waveforms and harmonics, and onto control voltage, filters and subtractive synthesis. You don't get to do your first patch until page 29. This thing is thorough and complete without being patronising. It's not 'just »

The Bullfrog's rear panel is home to 3.5mm sockets for CV and gate inputs and audio input, a quarter-inch audio output, a MIDI input and a USB C port.

The Bullfrog's manual is at the heart of the learning experience.

for kids'; this would be of great benefit to anyone wanting to learn about synthesis.

Peppered throughout are little quizzes which go back over what you've learned. They allude to answers and other resources online that are not in place at the time of writing. However, Girts assures me that these resources will be fleshed out once they've been pitched into schools. The other thing that's coming is a huge teacher's XL version of the Bullfrog. It's impressively large, I would say a bit bigger than a full-sized ARP 2600, but with all the knobs and switches scaled up. It has a built-in oscilloscope, stereo speakers and uses full-size jacks. You can imagine the teacher using this at the front of the class while the students follow along and experiment on their own standard Bullfrogs. I can also imagine how the room would be filled with the beautiful sound of experimental electronic music.

In Use

The Bullfrog squats confidently on your desktop and makes for a very comfortable place to play. The size of the knobs makes you wonder why we put up with Eurorack at all; this is how a synthesizer should be. It's all spacious and deliberate, easy to navigate and feels fabulous under your fingers. As you fill it out with patch cables, you never get the feeling that you're lost, hampered or caught up in it all.

It sounds great, like a decent monosynth should. The shaping on the pulse and sine waves is very welcome, and the noise adds to the versatility and provides the sample & hold with something to chew on. The mixer is brilliantly useful in combining waveforms and being clever with dividing oscillations, while the doubling up of CV inputs to both the filter and the VCA gives you the maximum amount of variation and modulation. The filter has plenty of bite and not too much drop-off as the resonance starts to overwhelm it. Including a delay was a great move. It adds interest and depth to what you're doing, although it's not voltage-controllable.

There are two attack/release envelopes, and they do a fine job. However, because there's no decay option, you do find yourself duelling the release knobs so the filter sweep can be heard before the VCA closes. On the other hand, they transform nicely into LFOs, have both uni and bi-polar outputs, and if you plug a cable into the gate input they become independent of whatever you're playing.

I'm not going to say the Bullfrog is a fantastically innovative synthesizer, but it's interesting in the way a Moog Mother-32 is interesting. It's designed to be foundational and straightforward, and so you will inevitably exhaust the monosynth bass line vibe. However, with a bit of experimentation and a combination of sample & hold, clock division, the wave shaping, delay and the mixer, you'll find a surprising amount of creative and tonal possibilities. Of course, you can also spill out into other modular synths and systems. But then, if we're talking about innovation, we really need to talk about the slot.

The Slot

The one bit of esoteric mayhem in an otherwise ordered synthesizer environment is the wide blue slot. The slot is for sliding in Voicecards. Voicecards can either expand the functionality of Bullfrog by adding components, or they can be used as a cool but labour-intensive method of patch storage. You may have seen something similar on Erica Synths' PICO System III, but Bullfrog takes the idea to a whole other level.

In the box, you get an Acid Bass Voicecard, a Sequencer, a Sampler/Looper and three blanks for your own patches. Forthcoming cards include an oscilloscope, a high-pass filter and an electric organ, which gives you a good idea of the range of possibilities.

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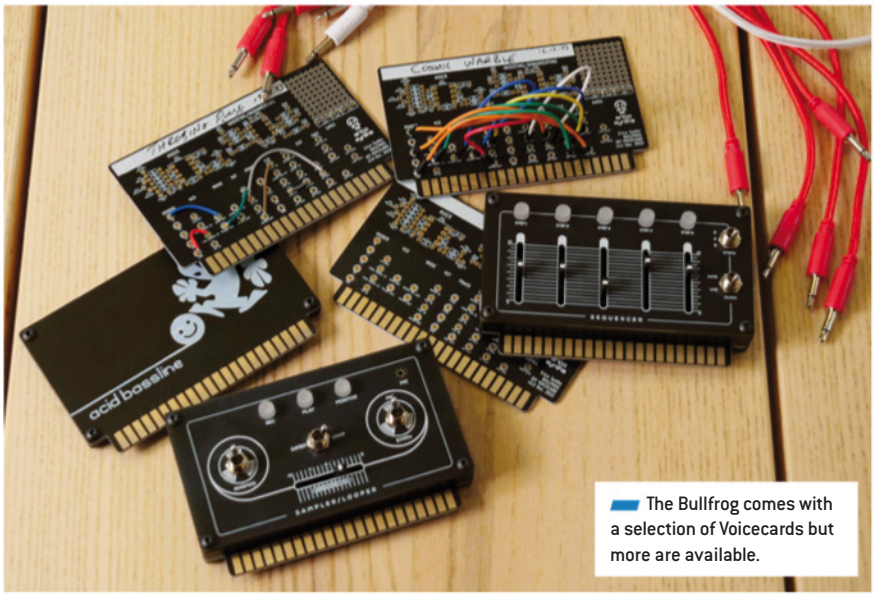
» The Acid Bass card is a straightforward patch that makes all the connections internally, removing any need for patch cables. Plug it in, and providing the knobs are pointing to something helpful, you should get a nicely resonant bass sound. The Analogue Sequencer keeps whatever patch you've made and adds a five-step sequencer. You've got sliders for pitch and buttons to turn the steps on and off. It will run with the sample & hold rate, or it can be step-triggered using a gate. The great thing about them is that you can plug them in, and you're off without having to know how to patch the synth together. While that may defeat some of the points of the synth as a learning opportunity, it is immensely satisfying. Any patching you do on the front panel will override anything on the card, so it's a great place to start experimenting.

The Sampler/Looper is a slightly different animal. The idea is that you can record up to 10 seconds of audio and play it back like a tape machine, or trigger it like a sampler. You can record from the synth or from a built-in microphone. The recording can be looped or gated, and you can shift it around in speed and pitch with a slider. It has a fabulously lo-fi and tape-style feel about it. It interacts with MIDI really well and manages a remarkable four notes of polyphony. On one switch you can engage a ring modulation effect that uses the Bullfrog sine wave as a modulator. It's all wonderfully mad and rough around the edges. I had some trouble getting it to work as I couldn't quite grasp how the sound was getting back into the synth. But after a lot of experimentation and reading the manual a few times, it clicked. The kids are going to love this.

The Bullfrog is fairly substantial, measuring 40 x 385 x 210mm and weighing 1.8kg.

If you like a bit of soldering, you're in for a treat. With a chunk of solder and some wire, you can more or less permanently melt your own patch onto the blank Voicecards. The way I approached it was to build a patch on the Bullfrog with cables, then copy the connections to the card. The only flaw in that idea is that not all the connections were there. In particular, the waveform Mix Out and the individual inputs to the mixer are missing, so you can't build a patch that combines waveforms.



The Bullfrog comes with a selection of Voicecards but more are available.

That aside, the satisfaction in soldering your own patch on to a Voicecard is tremendous, and you've got three of them. This really should be a standard across all synthesizers. The top half of the card has room for some chips and extra components. There are also power points on the board, which opens up all sorts of development possibilities to people who like to build circuits. I'll just stick to the patching.

Sticking patches on Voicecards and swapping them out is a wonderful way to engage with your synthesizer. They are hot-swappable so you can whack another one in right in the middle of a performance. There's huge potential for interesting developments if you want to dig in deeper.

Conclusion

It's difficult to separate the synthesizer from its educational intentions, but I don't think you need to. While it is deliberately simple, it will most likely teach you a thing or two about synthesis. The Bullfrog has a beautiful full-range analogue oscillator and a smooth resonant filter with enough functions to make it a useful addition to any setup. Its size and sturdiness make it a delight to use, and its looks would make it a fabulous coffee table talking point.

But it's the blue strip of possibility that really elevates Bullfrog into something a bit special. The patch saving is brilliant, and I could make cards all day. The included Voicecards brings playful and exciting things to the table, and there's no knowing where that might go.

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NICK ROTHWELL

Bitwig Studio 5.1

Music Production Software

Bitwig goes from strength to strength as v5.1 introduces a host of new features.

It must be pretty close to 10 years since I first visited Bitwig in their top-floor offices in Berlin, where they were showing off the first version of their Studio DAW. In the intervening time the software has acquired a powerful modulation system, a modular grid-based synthesis and effects environment, and much more besides. We are now at version 5.1 of Bitwig Studio, itself quite a step beyond version 5, so where has the modular music making program got to one decade on?

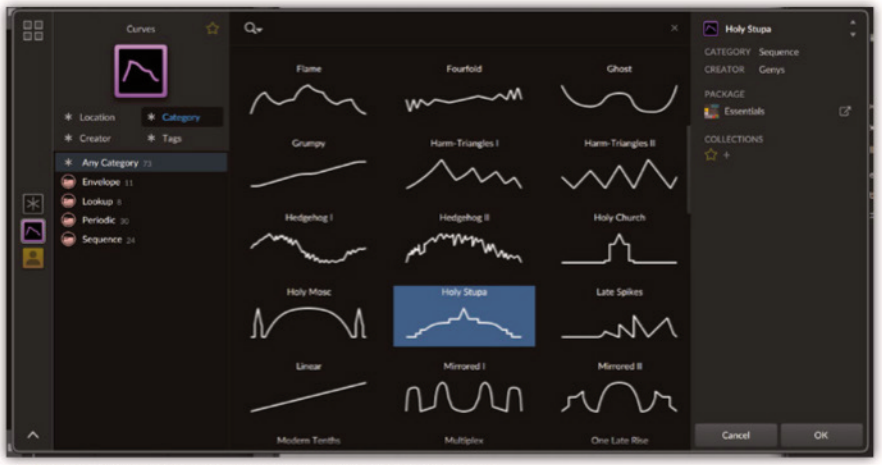
Curves & Segments

Let's jump straight into modulation. Bitwig 5.1 sports a new set of modulators, and modules for the Grid environment, that are built around an extensible library of freeform curves. Overall these components are referred to as MSEGs (multi-segment envelope generators). There's a browser for selecting curve shapes, and edited shapes can be added to the library at any time. (We'll look at editing shortly.) These curves are truly generic: they can be used as waveforms, waveshapers, LFOs, sequencers, envelopes or even keyboard tracking functions (should you think linear pitch-tracking is a little passé). The curves are categorised (envelope, lookup, sequence, etc) but these tags are purely advisory.

Let's look at the modulators first. Segments is an envelope supporting one-shot triggering, single-point hold or two-point looping (forward or ping-pong). The hold or loop points can be anywhere in the curve, though they do have to be edit points: you can't arbitrarily loop part of a segment. (You can add more points to a segment if you need them.) As with many modulators, Segments can be per-voice or monophonic, in which case an instance can retrigger per note-on, or trigger only

for a new note when none is already held. There's a smoothing parameter, with variable time, to take the hard edge off any shapes with sharp corners, or to create more gradual movement.

Curves is an LFO with tempo or Hertz timing, mono or polyphonic as per Segments, with sync'able phase, or the ability to randomise phase per note. Keytrack+ is probably the oddest of the bunch, using a curve to map incoming note pitch to a modulation value. There's already a relative keytrack modulator, which is fairly sophisticated, allowing a multi-break-point transfer function to be set up by MIDI note and level: this remains as a legacy feature. Keytrack+ allows arbitrary curves to be used instead. I'm not sure I'd need this instead of the existing keytracker, but some possibility may still come to mind, maybe involving some radical kind of modulation by distinct keyboard zone.



Hedgehogs to holy stupas: some of Bitwig's library curves.

Segments and Curves are both available as modules in the Grid modular environment, and Segments works as a drop-in envelope generator in the Polymer synth. Keytrack+ isn't provided directly in the Grid, but there's a curve-based waveshaping module called Transfer, which you can certainly attach to incoming pitch to achieve the same end. Transfer has a drive parameter calibrated in dB and an anti-aliasing option, suggesting that it's really intended for use in audio processing.

The Grid also uses MSEG curves to provide Scrawl, an oscillator module that plays a curve as a single-cycle waveform — again available in Polymer also — and Slopes, a data sequencer that locks to the phase of the Grid device (specified in bars or beats). Slopes isn't available as a modulator outside the Grid, but the Curves LFO modulator can be tempo-synchronised, and since it uses the same curve shapes it can serve the same purpose.

All modules and modulators based on MSEGs share a common editing interface. Click on any curve displayed in any of these modules or modulators, and a pop-up window appears showing the curve with its control points and a set of editing tools. Top-left are icons for selecting a curve from the library, and for saving an edited curve back into the library with a new name and metadata. (Each curve is saved in Bitwig's library in BWCURVE format, which at present is undocumented; it would be great to see third-party curve libraries, or tools to generate new sets of curves algorithmically from scratch.) Each curve sits in an editing grid which can be resized, and there are tools for altering curve segments in this grid as triangle or block-shaped elements, or you can turn the grid off for completely freeform editing. Alt-drag adds curvature to segments. If you're editing a Scrawl oscillator then the waveform you're hearing will change even as you drag points to edit, which is a nice touch.

The overall naming is a little confusing (Slopes, Segments and Curves are similar names for similar features, while Scrawl is the oscillator). The Grid also has a 'Curve' »

Bitwig Studio 5.1
$399

PROS

- New oscillators, filters, waveshapers and multi-segment modulators enrich the ecosystem of audio components available to the Grid modular environment and the Polymer synth.
- New effects devices for hosting filters and waveshapers.
- Modulators and remote controls can be added at track and project level, and track remotes can be laid out in the mixer view.
- New clip-launching modes and options add more versatility in live performance.
- Improved browser for devices, presets and plug-ins.
- Plenty of other nips, tucks, tweaks and fixes.

CONS

- Bitwig Studio increases yet again in power and sophistication, which means more time is needed to master the details.

SUMMARY

Bitwig Studio 5.1 is a solid upgrade, with a good offering of new filters and waveshapers for the Grid, more powerful modulation and remote control machinery, a better device browser and a host of other small improvements. Quite a few features have been reworked, so it'll take a while to acclimatise, but the improvements really make this upgrade a no-brainer.

» module for shaping, which is distinct from Curves. However, the Grid's default colour coding helps differentiate between the variants.

Track & Project Modulation

Every device in a Bitwig project has two sets of control inputs: the modulators and the remote controls. The modulators can be thought of as enhancements to a device, adding automation-like behaviours that might take effect note by note (for example, envelopes) or for the device as a whole (for example, traditional MIDI controller inputs or on-screen X/Y pads). The remote controls section supports individual pages of up to eight controls; each control is really just a convenient duplicate of a parameter in the device or one of its modulators. Because the controls in a page are in one place they can be conveniently mapped as a group to a physical MIDI controller (such as my cheap but trusty eight-knob Akai LPD8). A device instance can sport multiple pages of controls, and by default the live mapping of physical MIDI knobs to on-screen controls follows the currently selected device and page.

So far, nothing new: Bitwig has had modulators and remote controls for a while, although Bitwig 5 now gives most device modulators their own remote control pages alongside those of the device or the specific preset. The major enhancement in this area is that tracks themselves, and a project as a whole, can sport modulators and remote control pages. Group tracks can also have their own modulators and remote controls, targeting the group itself or the tracks within the groups.

Whether or not you have a physical MIDI controller that you are using to operate the remotes, it's a bit of an inconvenience to have to click between tracks and devices to navigate between control pages. Now it's possible to show a panel of controls directly in the mixer's track view. Do this with the controls set up at track level, and you can see them across all the tracks at a single glance. As you navigate between tracks (using the computer's arrow keys, perhaps), by default the MIDI controller assignment will follow the panel of the selected track. Set up your control mappings correctly, and you can probably lay out all the control points for an entire live set across the track view, all visible and reachable at once. As an aside, it's possible to MIDI-map the selection of the pages within a remote control panel for a device or track, and the pages do indeed change in the mixer section as well. One feature request: some kind of textual indication of the actual controls page currently showing in the mixer.

The Segments modulator and the pop-up editing interface for MSEG curves.

This is all very well for remote controls at the track or project level (the project controls are associated with the master track), but you're equally likely to want access to instrument and effect parameters from the mixer. You can do this manually by assigning a mapping from a track control page to a device control page. The track control panel will also display some device pages directly, as 'aliases'. Quite which pages are available is a bit of a mystery, but the target usually seems to be the last instrument in a device chain; effects devices aren't included. (Bitwig have since confirmed that instrument preset pages are made available as aliases, for one instrument at most per track; global instrument and effects pages are not.)

Back to modulators. The modulators at the track level cannot operate polyphonically as they are not attached to instruments, which can layer voices — so the voice stacking modulator, to pick a contrived example, is disabled. LFOs can be run in monophonic mode but not per voice. I was surprised to discover that the note-counting modulator, which provides a stepped modulation output as it counts incoming notes, will actually work at track level, as will various MIDI-in modulators like the mod-wheel-driven vibrato LFO.

Track modulators can target any device in the track, and in the case of a group track, any device in the tracks that are grouped. In addition, track modulators can affect the track themselves: level, effects sends, pan and mute status, and crossfade mode. The modulators at project level can affect all tracks including the master, plus global settings such as fill (used to control the note triggering operators) and tempo. Tempo modulation has no visual

Remote controls at the device, track and project level.

feedback (there's no readout of the current effective tempo), and, as fun as it might be, I'd recommend steering clear of this: the ramifications to things like audio warping in recorded clips are unclear, at least to me. (Examining a Bitwig session with an LFO affecting tempo, I saw the real-time position indication in minutes and seconds was also changing speed, which was slightly disconcerting.)

Filters, Waveshapers & Bite

There are four new filters and six waveshapers added to Bitwig's audio toolbox. These are implemented at the level of the modular Grid environment, so can be used directly if you're building or modifying Grid instruments or audio effects. The filters can be swapped into Polymer, the Grid-built synth, and both filters and waveshapers can be swapped into Filter+ and Sweep, two new effects devices we'll look at shortly.

There are now sufficiently many filters and waveshapers (10 and 14 respectively, by my count) that they are divided into categories. Filters are now 'structural' for the basics, 'inspired' for those that nod towards classic hardware, or 'character' for something new (in concept, and new in Bitwig 5). Waveshapers are 'one knob', which speaks for itself (although these ones also have an anti-aliasing switch), 'parametric' for more sophistication or, again, 'character' for something new.

Remote control pages reformatted and slotted into the mixer view.

The Browser

The content browsing machinery in Bitwig 5 has been given an overhaul. For a start, device and preset icons in the browser have a bit more space between them, and the labels to indicate plug-in type are now big enough to be readable without having to squint. This sounds like a minor change, but browser icons appear in quite a few places in Bitwig's visual interface, so the overall effect is to make the program feel a bit less cluttered.

The top of the pop-up browsing tree is at a library level, providing access to packages, from which you can drill down through multiple views available: location, file kind, category, creator, device, tags. These just rearrange the presentation of the assets, although 'tags' lets you apply tag filters. If you ignore the packages you can click the asterisk for 'everything' — again with multiple views available — or use the left-hand icons to filter by instruments, audio FX, note FX, plug-ins, and personal library. Instruments are themselves categorised by sound style (drum presets, keys, synth, bass, etc). There's quite a lot going on here, some of it subtle — the tags filtering has two levels, for example, and some browser sub-panels contain filters which influence the main panel. While the browser was good for drilling down while looking for particular types of preset, it broke down a little when looking for specific instruments or more abstract elements such as device racks. Luckily you can still type in the name of the thing you're looking for, the procedure I adopt 99 percent of the time.

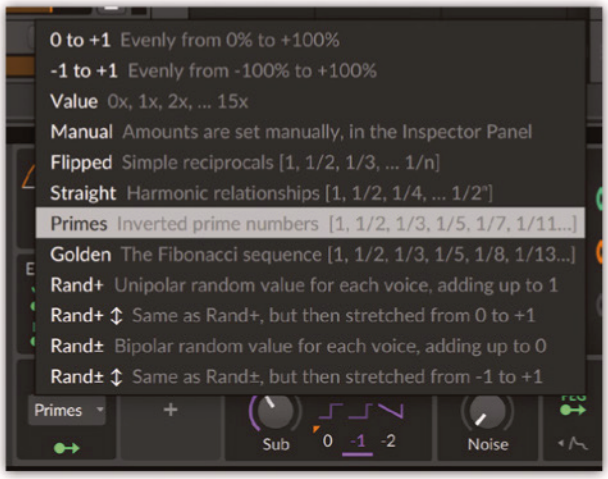
I used a basic preset for Polymer to work through the new filters. 'Vowels' is a newcomer in the 'inspired' category, dropping in alongside the existing Oberheim-style XP and Moog-style low-pass MG. To say that Vowels is distinctive would, I think, be a bit of an understatement. It can dynamically blend across five slots, each of which carries one of 27 possible vowel sounds. Then there are selectable voice profiles: different variants of male, female and child. There's a resonance control, detuning of the filter bank, and even a variable topology to rearrange the audio flow through the filters. To add to the potential for variation, all of these parameters can be modulated. The Vowels sound is rich and clean, but you might want to steer clear of the most obvious vocal-like treatments, unless that's really your thing: luckily, there are enough controls here to get into more interesting and synthetic, and much less vocal, territory.

The 'character' filters, called Fizz, Rasp and Ripple, are variations on a theme: there's an internal feedback connection, and associated feedback gain or cutoff controls. The Ripple filter has a 'Nature' control with three settings: Earth, Wind and Fire. (Perhaps Water was too difficult to implement?) The documentation for these is a little bit inscrutable, but that really doesn't matter: they sound wonderful, picking out and tickling the harmonics with some gentle distortion, without things getting out of control. Ripple, in particular, has a way of bringing out details in beat patterns, which I found particularly engaging.

Not being a modular synth hardware geek, I don't spend a lot of time with waveshapers, so these new 'character' modules didn't jump out at me, but the Howl and Shred variants seemed the most interesting in terms of pulling interesting things out of the frequency spectrum, and despite their names seemed eminently controllable and predictable. All of the waveshapers seemed to be good complements to wavetable oscillators, adding an extra dimension of continuous tonal exploration.

Outside the Grid, there are a couple of new audio effect devices that act as hosts for these filters and waveshapers — much like the Polymer synth, these are actually built using the Grid, so you can open them up and dive deeper into the details if you so wish. Filter+ has a slot for a waveshaper and one for a filter, the audio passing through them in that order. For good measure it also has a dedicated LFO, which can run at audio frequency for generating side-band harmonics, and a modulator which tracks audio level. The LFO runs in stereo, with a phase offset between left and right channels: this is indicated in the LFO icon. It's not immediately clear how this affects the operation of Filter+ — the presets didn't present any obvious stereo behaviour — but if you convert the device into an FX Grid instance then you can use »

Waveshaper and filter sit side by side in the Filter+ effect.



A plethora of voice stacking distributions.

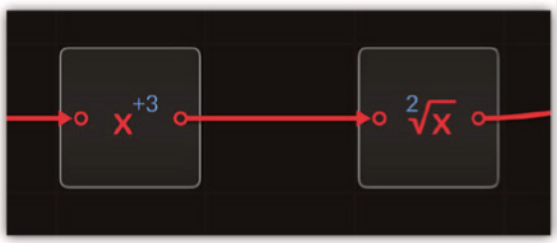
» the stereo output from the LFO however you like. There's no editing of the LFO phase or shape skew in the Filter+ panel, but the Grid view provides access to these.

If Filter+ isn't sophisticated enough for your needs, a device called Sweep hosts two filters with a waveshaper in the middle, and has a morphing control to shift smoothly between different variants of serial and parallel routing through them.

While we're on the topic of modules for the Grid (and for Polymer), there's a new oscillator module called Bite. It's a dual-oscillator module: the identical oscillators come with a selection of wave shapes, most of them amenable to PWM. Oscillator B can be hard-sync'ed to A, but more interesting (in my opinion) is that B can be frequency-modulated from A, and the mixer output includes a level control for the ring-modulation of A and B. In the context of the Grid, the two oscillators can be decoupled so that they can be modulated by the left and right components of a control signal (a reminder: the Grid runs end-to-end in stereo).

I found that Bite is indeed capable of biting, and it took me a while to get some engaging FM and ring-modulated tones out of it (turning off hard sync helped!). In the end, I was more taken with the slow, sustained textures I was able to generate — helped a lot by the stereo detuning parameter, which sounds rather sweet. Bite doesn't contain any internal modulation controls — there's no dedicated LFO, for instance — so in the Grid you'd need to wire up your own, and in Polymer make use of the modulator slots.

The sheer selection of filters and waveshapers, and their parameters, did start to feel a little overwhelming, but Bitwig 5 comes with a new package called Wavescapes and Filter Shapes, which is packed full of instrument and effect presets showing off what can be achieved.



Some mathematical processors in the Grid.

Voice Stacking & Control

The instrument voice stacking features have been expanded, shifting them from the category of cool into slightly mind-bending. The voice stacking limit, which used to be five in Bitwig 4, is now 16. The Stack Spread modulator, which applies a distinct degree of fixed modulation to each of the layered voices, has been expanded with a range of distributions. The inspector panel for the modulator has a visual per-voice visualisation indicator, which shows how much modulation has been applied to each voice: these indicators will be static for most distributions, but for the random distributions the indicators will jump to new positions for each note played. If you really feel that you need even more voice-specific control, the new Voice Control modulator provides a distinct modulation source for each voice, so that each of the layered voices in a stack can modulate a completely different set of device parameters than any other voice. So, yes, the voices can be manually detuned or panned — or they can be playing different waves in a wavetable, have different filter algorithms (so yes, you presumably can get Earth, Wind and Fire from playing one note through the Ripple filter), different envelope shapes, and so on. I'm sure there are even more wild and bizarre possibilities that I've not thought of.

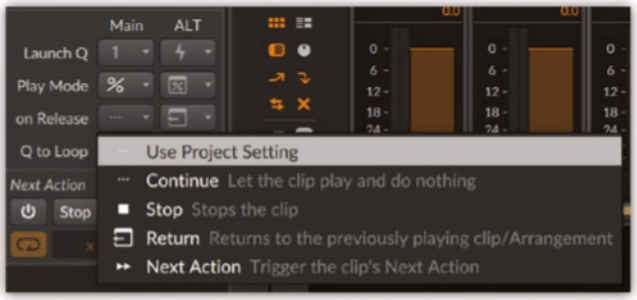
In the Grid, there is a Voice Stack Mix audio module offering per-voice mixdown. I had to sit down and think about this one for a moment, and the documentation is a little terse, but my take is: drop this into the audio path of the Grid instrument at some stage, and it will apply level and pan on a voice-by-voice basis: there's a control panel that lists the voices with their own volume and pan knobs, and mute and solo buttons. All the controls in this panel can also be modulated — this appears to be averaged from all stacked voices monophonically. There is also a simpler Voice Stack Tog module with buttons for enabling or disabling voices — and again, these buttons can be modulated.

New Grid Modules

A number of new modules have been added to the Grid, over and above the various MSEG modulators and shapers. There are some additions and updates to the maths modules, including exponent and root (the exponent can be negative to divide by a value), a pitch operator for imposing an S-curve on a signal or phase, a signal amplifier, and converters between note pitch value and Hertz. There is also a wavetable LFO, index-sweepable, using the same tables as the wavetable oscillator. It took me a while to find the modulation input for wavetable position: it's towards the right of the panel, rather than along the left edge.

Clip Launching

The clip launching machinery has been upgraded, allowing more flexible performance options. The most basic clip launching method ('Trigger from Start') is, unsurprisingly, to launch from a clip's start. Depending on the quantisation setting for the clip, this might be immediate (unquantised) or at some bar/beat interval in the future. 'Legato from Clip (or Start)'



The various clip launching modes.

MIDI mappings for navigating scenes and tracks.

looks to see whether another clip in the track is playing, in which case playback 'punches in' at the corresponding position in the new clip. (The starting position seems to be the mathematically sensible point if the two clips are different lengths, wrapping the position if necessary.) 'Legato from Clip (or Project)' behaves the same, unless there's no clip playing, in which case the punch-in position is calculated from the transport position, wrapped round by the clip length. Another way to think about that is that you have the option to launch the clip immediately without quantising, but its playback position will be quantised according to its loop length. 'Legato from Project' does this punch-in calculation from the transport regardless of whether any other clip is already playing.

It is now possible to perform an action on release of mouse or trackpad button (or controller pad). Apart from the default action of doing nothing (and letting the clip continue, if it hasn't stopped already), there's stopping the clip (which respects quantisation) returning to the previous playing clip (or the linear Arrangement), or triggering the clip's Next Action.

A clip can have two completely different groups of settings for launch quantisation, play mode and release action. One (called Main) is the default, the other (referred to as Alt) is enabled by holding the Alt/Option key on the computer, or dedicated Shift key on a controller. The actions available for Main and Alt are the same, and you can even pull tricks like Main-clicking a clip and then Alt-releasing it, or *vice versa*.

All these settings are available on a clip-by-clip basis, but there is also a set of defaults at the project level if the settings need to be shared and managed more centrally. Also, these settings can be established for a scene, optionally overriding the settings for the clips in the scene. (Arguably, it would make sense to have defaults at the track level too.) I was rather pleased to discover that scene launching also respects release actions: 'release' a scene launch and the release actions of all of its clips are activated.

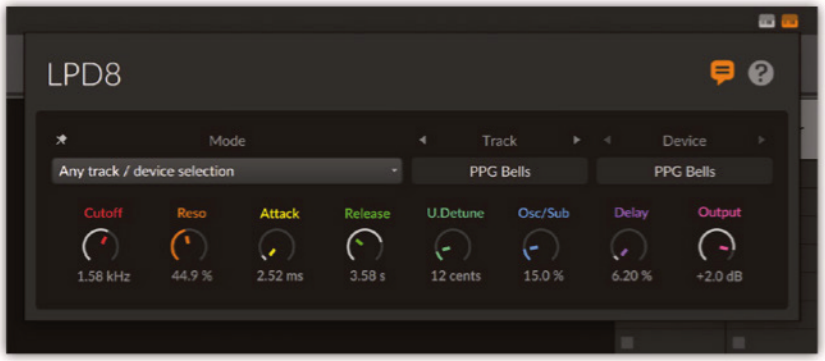
It's possible to MIDI-map clip and scene launching, and I can report that 'release' actions work as well: note-off acts as release, and if a continuous controller is mapped, then a high value is a launch while a low value is a release. As I write this, there is no way to generically map a MIDI controller to select between Main and Alt actions; that action has be to supported in controller scripts for specific controllers (currently, Novation Launchpads and Akai APCs).

MIDI mappings can be set up for specific scenes, but there's also a set of mappable controls for navigating the scene list as a whole: mapping targets to launch the previous or next scene, or to 'cue' the next scene to be launched. It's also possible to MIDI-launch a track: this launches the clip, or clip slot, for that track at the cued scene. Perhaps for simplicity's sake, these controls don't invoke release actions. It's certainly nice to see more sophisticated scene launching options, which in combination with the remote control views in the mixer should make live performance more flexible and less onerous to set up.

Controllers

There is a bit more visibility of MIDI controllers attached to a Bitwig session, and which mappings are active. All attached controllers are denoted by small icons at the right-hand end of the application title bar. Click on one of these and you get access to controller parameters. I didn't get much information about my Arturia KeyStep (which, while providing a decent keyboard, sports no usable knobs or sliders), but my Akai LPD8, added as a generic eight-knob controller, showed up fine. By default a controller follows the currently selected track and device — and within the device, the current control page. It's possible to filter what kind of panel the controller can be connected to: devices, tracks, the global project pane, or all of these. It is also possible to lock the controller to a particular track or a particular device. However, a panel can have more than one pane of knobs, and there's no selection or navigation here between panes: you can only change pane in the device itself, or via a separate, manually configured MIDI mapping.

This drop-down panel would appear to be performative, at least in intent. There are little arrows to nudge back and forth along tracks and devices, though it won't navigate into grouped devices or chains inside devices. (It will navigate nested tracks in groups though.) There doesn't seem to be any way to MIDI-map these nudging arrows. It would also be rather handy if this panel could show manual MIDI mappings for tracks and devices alongside the controller-level ones — the former are currently shown for the entire project in a single mappings panel.



Using the knobs on a generic eight-knob controller.

Conclusion

When I started working with Bitwig Studio 5, and then version 5.1, nothing really jumped out at me as the Big New Idea justifying the upgrade. Yes, the browser is markedly better, and the interface overall is cleaner and clearer. It took a somewhat closer look to get a sense of where the builders had been at work: Grid filters and waveshapers, modulators, clip launching, remote controls, track controls, and so on. A lot of effort has been applied here, removing pain points from the previous version, and really adding power and flexibility where it's needed. More and more, Bitwig is feeling less like a DAW and more like a sophisticated modular toolbox for sound design and performance. The actual box hasn't changed that much, but many of the tools have been improved, some replaced, and a whole bunch of new ones added. I spent a considerable period of time while Bitwig 5 was in beta going back to using the Bitwig 4 release for commercial projects, and I'm glad I can finally shift up to 5.1 and adopt the new ecosystem in anger.

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JZ Microphones BT202

Small-diaphragm Capacitor Microphones

JZ's first small-diaphragm mics combine a distinctive appearance with excellent technical specs and an appealing sound.

NEIL ROGERS

JZ Microphones have been building an excellent reputation for their distinctive-looking mics for well over a decade now. Manufacturing in Latvia, they aim to offer high-end sound at a more manageable price than most established European marques. The most well-known JZ models can be found in their Vintage series, which includes a selection of large-diagram capacitor mics that promise a classic valve-like sound from solid-state electronics, and which employ their unique Golden Drop capsule design. I had the pleasure of reviewing the Vintage 12 from this series last year, and had plenty of good things to say about how it fared in my studio: www.soundonsound.com/reviews/jz-microphones-vintage-12.

The subject of this review is JZ's first small-diaphragm capacitor mic, the BT202. Sold exclusively as a stereo pair, it's intended as a high-quality option for all-round instrument recording.

Pencil Power

The BT202 is a smart and compact-looking mic and, like all JZ designs, has a characterful appearance that doesn't immediately conjure up associations with a specific design of yesteryear. The technical specifications are impressive, with dynamic range quoted as 128dB and maximum SPL at 140dB, meaning the (A-weighted) self-noise figure is only 12dBA — about as good as it gets for a small-diaphragm mic. The published frequency response is flat from 30Hz up to 6kHz, before a slight bump peaks around 8-9 kHz. The BT202s have a fixed cardioid polar pattern, and the same Golden Drop technique JZ use in manufacturing their large-diaphragm capsules is employed for the 13mm capsule found in this model. JZ say this makes the diaphragm lighter

and more responsive to transients. The electronics in the BT202 are all Class-A, with a transformerless circuit that is intended to offer an uncoloured, faithful representation of the source.

JZ are one of a number of mic manufacturers who offer the option of purchasing a proper wooden case for their microphones at an additional cost; as standard, the BT202s come in a cardboard box. I totally get the reasons behind this, but I think the price point here is very close to the level where a proper case should be expected as standard. That minor grumble aside, the BT202 is a stylish and 'funky'-looking mic that nevertheless manages to maintain the air of a serious, businesslike studio tool. I also liked that the slightly textured, matte finish makes these lightweight mics more tactile and less likely to be dropped by chunky-fingered engineers.

Riding High

The first port of call for the BT202s in my studio was as a pair of drum overheads. I was curious to see how they performed in this role, as my live room seems to naturally present cymbals quite brightly, meaning I often favour large-diaphragm capacitor or ribbon mics above a kit. I was impressed with how they sounded: the resulting capture of the cymbals was indeed bright, but it felt very honest and was very easy to rebalance if necessary when mixing. I used the mics on a few different drum recordings over the review period, and appreciated how they delivered a nice sense of 'left to right' when positioned above the kit, broadly focused on the hi-hat and ride cymbals at either side. I also liked the fact that the overhead sound they gave me was on the leaner side and thus sat more naturally with the close kick and snare mics, which provided the required body and weight when needed.

I record a variety of acoustic instruments at my studio, and acoustic guitar in particular is an instrument where I would naturally gravitate towards small-diaphragm capacitor mics. The BT202s didn't disappoint, and I was struck again by how honestly the mics captured what was in front of them. These are not mics that will flatter a poor-sounding guitar — or player! — but with careful placement they produced detailed, nicely balanced acoustic guitar recordings. I could often hear the subtle lift in the frequency response toward the very top of the midrange, but it nearly always felt like it was working in my favour, gently pushing forward the articulation of the pick or fingers hitting the strings. One other thing I noticed when recording acoustic guitars is that the BT202s have quite a strong proximity effect. A degree of bass tip-up is, of course, inevitable with any cardioid mic, but I seemed to be a touch more aware of it with the BT202s than with other similar mics in my collection.

The BT202s were already proving to be highly capable all-round mics in my studio, then, but when I tried them on my upright piano, I was suddenly in the 'highly impressed' category. I like to remove the front panel of my Knauss upright when recording, and I positioned the mics so that they were pointing toward the strings at a distance of around 30cm, at either end of the range where the musician would be playing. Sitting in from of my monitor speakers to listen, I had a wonderful sense of the piano moving between my left and right speakers, which quite simply brought a smile to my face. It wasn't just the stereo imaging that was pleasing, though: it was also the transient response and how nicely the top end of what is quite a dark-sounding piano was gently enhanced. This was quite possibly the best I have heard my studio's piano sound in the seven or eight years it has been there.

"The BT202s were already proving to be highly capable all-round mics in my studio... but when I tried them on my upright piano, I was suddenly in the 'highly impressed' category."

It didn't stop there, either. I also used the mic for recording a violin, positioned about a metre above the player's shoulder, and it again produced a detailed, balanced capture of what can be a very unforgiving instrument to record. One final, surprising role in which I liked the BT202 was as a vocal mic for a female folk singer who I record quite often; I loved how the subtle 8-9 kHz bump brought a little sparkle to her vocal. I also liked how the leaner low end felt ready to go straight into the mix, and I'm now wondering why we don't use mics of this type more often in vocal sessions!

To Sum Up

Many engineers complain that modern small-diaphragm capacitor mics can sound a bit bright and unflattering — which, perhaps, partly explains the ridiculous prices of certain vintage models. The BT202s come across as slightly darker and softer than many mics I've heard in this family, which suited my ears and requirements very well. I have other mics that give me a bit more 'zing' or 'air', but they often end up sitting around not getting used for long periods, as they only work in certain settings. As all-round studio mics for drums, guitars and any general instrument recording, the BT202s just got on with the job of being quiet, accurate microphones that you can work with to get clean recordings, or pair with a more colourful preamps for a bit more of a vibe. And if you're looking for a dedicated pair of piano mics, the BT202s should definitely be on your shortlist. With the caveat that I only tried them on my own upright piano, they worked superbly! ■■■

Listen Online

To hear the BT202s in action on some of the sources described in this review, surf to https://sosm.ag/jz-bt202-media.

JZ Microphones BT202
$999

PROS
- Versatile, smooth-sounding mics with just a hint of high-frequency emphasis.
- More affordable than most European-made alternatives.

CONS
- A proper box is a cost option.

SUMMARY
The BT202s are high-class yet reasonably priced pencil mics that will give good results on almost any source.

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Teaching Machines Wellspring

Dual-channel Spring Reverb

Spring reverb can be a one-trick pony, though it's a pretty good trick! This one incorporates additional processing to make it more versatile.

NEIL ROGERS

Spring reverb seems to have enjoyed something of a renaissance in recent years. While I'm generally very pleased about this — I've long been a paid-up member of the spring 'verb fan club — it means that well-regarded vintage units now command much heftier sums than when I acquired mine! So I'm always interested when something new comes out, and particularly when it's not just copying an old unit. The latest such device to grab my attention is the Wellspring. An all-analogue design, it's described by its manufacturer, the small Welsh company Teaching Machines, as "a stereo spring reverb with lots of cool complimentary [*sic*] features".

At the heart of the Wellspring is a pair of 15-inch spring reverb tanks, which are housed in a metal case that's been given a vintage hi-fi aesthetic. The included rack ears and rubber feet can be fitted or removed by the user, so it can be bolted into a standard 19-inch studio rack or sat on a desktop as you prefer — a nice touch. The build quality seems very good to me, with all the controls feeling solid and reassuring to use. Equally reassuring is that Teaching Machines offer a five-year warranty as standard.

On the rear, quarter-inch jacks provide the unbalanced stereo inputs and outputs, as well as a ±5V inlet for control signals, and there's a high-impedance jack input on the front to cater for guitarists. The unit has input and output level controls at either side of the front panel, and there's a small 'clip' indicator light near the switches for power and for engaging the guitar-based input.

Effects, Filters & Magic

Spring reverbs always have a strong sonic character, and because of that people tend either to love them or find them limiting, in any given situation: it's

often either just right or plain wrong for a particular source or mix. To make them more versatile, many manufacturers back in the day built some kind of EQ or wet/dry controls into their designs, but while this did help, the user was usually still left with limited control over the reverb's decay. Notable exceptions were the higher-end AKG models, which employed some complicated techniques for adjusting the length of the decay time, or 'tail' as it's known. The Wellspring doesn't recreate the AKG approach, but it does attempt to tackle the same issue in a different way: it features a range of analogue effects, including a pair of delay lines, and four different filter/EQ settings and modulation sources that act on the delay and filter. Taken in combination, these allow you to shape the springs' natural decay and character. There are also creative options for introducing feedback to parts of the circuit, and a few more clever features that I'll explore in more detail below.

Broadly speaking, there are three different sections with which you can experiment, and I'll take you through these as they appear on the control panel, from left to right. First up is a pair of delay controls called 'time 1' and 'time 2'. These can either be used independently, to introduce different delay times on the left and right channels, or be linked so that 'time 1' controls the delay for both channels. The feedback control works the way you might expect, and can do anything from introducing a single repeat up to 'infinite feedback'. There's also the option of swapping between 'parallel' operation and a ping-pong-style effect, in which the left and right channels are inverted as they pass through the feedback loop. I'll discuss this in a bit more depth later, but for now let's just say that I reckon the designers have undersold this section of the unit somewhat in their literature – it's a great-sounding analogue delay in its own right!

Teaching Machines have been far keener to highlight how effective their modulation features can be in 'calming down' some of those unpleasant ringing effects that, as I discussed above, are often heard in spring reverbs. They have a good point, too. The Wellspring offers two broadly different flavours of modulation, with the option of switching between sine or sawtooth waveforms, but if you want something more complex, you can always patch an external control source into the rear-panel socket. There's also the option of inverting the left and right channels to allow the modulation to be in or out of phase.

My experience of using the Wellspring suggests that the most important control in this section is the modulation speed, and this has a range of settings from super-slow to very fast. To aid your ears, a small yellow LED flashes in time with the modulation speeds. This can be very helpful, although when I positioned the Wellspring in front of my mix position, as I often do when evaluating devices for review, I found it a touch unpleasant on the eyes. That should be less of an issue if set off to one side, or in a rack, of course. The simplified block diagram included with the unit is essential reading if you're going to get your head around how things work, and this shows that the modulation is applied simultaneously to both the delay and filter sections of the circuit.

"The Wellspring features a range of analogue effects, including a pair of delay lines, modulation options and four different filter/EQ settings that... allow you to shape the springs' natural decay and character."

The Wellspring's EQ and filter options are powerful tools for shaping your signal before it hits the springs. As well the expected low- and high-pass filter options, you can choose a band-pass filter to only allow frequencies in a certain range to pass through, or a notch filter to remove a narrow range of frequencies. However, you can't use more than one of these filter types at once. A further, minor niggle is that you can't easily bypass the filter section, other than by selecting the high-pass option and turning the frequency control all the way left. This can seem a bit clunky when trying to fine-tune a reverb in a mix, and a dedicated bypass switch would have made it easier to assess things. Nonetheless, this section can have a dramatic impact on the effect in the context of a mix.

There are two 'wet/dry' controls, one at the end of the filter section and another just beyond. The first is for the filter and delay section, while the second blends between the output of that first wet/dry control and the direct output from the springs.

The last 'section' at the end of the control panel is the mysterious Magic, which has a single control labelled Intensity. What this does is send some of the reverb signal back into the delay, but with the left and right channels swapped. Not surprisingly, this can introduce some pretty crazy feedback-style effects. It can become unwieldy, but it's aptly named nonetheless: if used subtly you can achieve an effect that's very difficult to convey using words other than 'magic'!

A Reviewer's Tail

I wanted to start my adventure with the Wellspring by getting a sense of what the reverb sounded like, without bringing any of the additional bells and whistles into play. In what became a common theme during my early experiments with the unit, I had to refer to the block diagram to figure out how to hear just the sound of the springs. But once I felt sure that setting the wet/dry control fully dry would achieve this, I was suitably impressed. The twin reverb tanks provide a deep, rich-sounding spring reverb, with a long tail that decays in a smooth, natural and very satisfying way indeed. Subjectively, »

Teaching Machines
Wellspring
$1500

PROS

- Great-sounding stereo spring reverb.
- Filter section allows for effective shaping of the reverb tail.
- Analogue delay sounds fantastic with and without the reverb.
- Lots of creative options that will grow as you learn and experiment.
- A great 'new' alternative to vintage effect options.

CONS

- Takes a little while to get to grips with.
- The nature of the all-analogue design can be limiting at times.

SUMMARY

A wonderful-sounding, all-analogue stereo spring reverb. The filter, delay and modulation sections allow for flexible shaping of the spring effect, as well as forming creative effect options in their own right.

The Wellspring supports stereo and mono-stereo operation, as well as accepting external control signals for modulation of the delay and filter sections.

» I'd say it sounds more 'full range' and 'hi-fi' than my own AKG BX5.

This was an encouraging start, then, but as with most of my reviews, I was also keen to throw the Wellspring into use on a real session. In the first instance, I thought I'd try patching it in for use as a fairly conservative, general reverb option for an EP I was mixing for a lo-fi indie band. This worked, in that the depth provided by the Wellsping was an absolute delight when applied to vocals and a few select instruments. But the naturally long reverb tail was too much, and needed shaping to make it sit nicely in the mix. I know from experience how effective EQ and filtering can be in this scenario, but I first wanted to get a feel for how the modulation section might help. Happily, it did. I struggle a little to articulate exactly how or why, but Teaching Machines explain that it can have the effect of 'scattering' more dominant parts of the reverb tail, and I certainly had a sense of this being the case. Higher modulation speeds in particular seemed to calm down the bouncy 'doink' effect that resulted from more percussive elements going into the reverb.

After getting a sense of how the modulation could help, I turned to the filter section and it was here that I found what I needed for this particular project. All four filter options are highly usable, and my only real frustration was that I could only use one of them at a time. For settings where the reverb is more prominent, the low- and high-pass filters work superbly for gently shaping the effect away from the top or bottom of your mix. I especially like filtering out much of the top end so that the reverb acts as a kind of low-level 'thickener' on a mix. The band-pass filter works for using the effect in a more targeted way, and the 'notch' option worked especially well for settings where you want to just reduce a certain element of the 'springiness' — say on a snare drum or sibilant vocal. On the whole, the combination of the modulation and filter sections can work very nicely.

More Than A Reverb

I carried on using the Wellspring as a fairly conventional reverb on a few projects, and as I got more attuned to the ways in which the different sections interact, I found it to be more and more flexible and more and more fun. It was when I had some suitable downtime, however, that I was able to fully explore this device, and appreciate that it's much more than 'just a spring reverb'.

My initial suspicions were that the delay section might be intended as a sort of pre-delay to introduce before your reverb. You *can* use it like this if you wish, but it's also a simple but great-sounding stereo analogue delay in its own right. In fact, I often found myself using the Wellspring first and foremost as a delay when working with guitars or vocals, and then blending in the springs to taste. Having separate controls for the left and right channels is great for adding subtle (or even more extreme) interest to the delay effect, and the feedback control offers you a nice range of options for how prominent the delay should sound.

So if you like to sit, experiment and lose yourself with a piece of equipment, playing with the delay and modulation settings on the Wellspring should be well up your street. I invited more than one guitarist to work in this way, plugging straight in via the front input and getting creative, and they all enjoyed it — it's one of those bits of kit that encourages you to play and experiment not only with the effect itself but also with your instrument and playing technique.

Summing Up

There's definitely a learning curve if you're to get the best from the Wellspring, but the effort is richly rewarded, and hopefully this review will make that journey a little easier. This really is a great-sounding analogue effects unit. The springs themselves sound wonderful, and whilst you might not have a means of adjusting the length of the reverb tail directly, there's a fantastic range of features that can help you shape the reverb to suit any source or mix. These are much more than add-ons; rather, they're fundamental to the designer's intention of how you should use the Wellspring, and they allow you to create dark, characterful delays and modulation-style effects to your heart's content.

Hardware effects such as this will be seen as a luxury to many, but the price is pretty good for what's on offer — especially if you compare it with vintage spring reverbs. If you're on the look-out for a special effects unit in the price range of a vintage tape echo or, say, an AKG BX5, I'd strongly encourage you to put the Wellspring on your shortlist.

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Boss RV-200 Reverb Multi-mode Reverb Pedal

Boss's 200-series pedals strike a good, practical balance between the their simpler compact pedals and the larger, more sophisticated 500-series pedals. The latest addition to the 200-series is the RV-200 Reverb, which offers 12 different reverb types, including a new algorithm called Arpverb. The pedal can run from either three AA batteries (included) or an optional 9V power supply. Stereo inputs and outputs are supported, as well as mono, mono-to-stereo and stereo-to-mono.

It's a digital pedal, of course, and this one has 32-bit A-D/D-A conversion, while the reverb is generated by a DSP using 32-bit floating-point processing running at a 96kHz sample rate. All the expected reverb types can be found here — there's a Room, a Hall, a Plate and so on — but there are also some lovely 'special' reverb treatments too. These include Shimmer, the slow-attack SlowVerb and the new Arpverb, which imposes a shifting sequence on the reverb tail.

There are 127 memories for saving patches and as well as the dual onboard footswitches, which can be assigned to different functions, there's support for external switches and pedals if you need additional real-time control. Directly from the front panel you can access four memories and one manual setting, independently of any footswitch settings. MIDI I/O is supported on mini TRS jacks, needing just an adaptor cable to interface with five-pin MIDI gear, and Program Change messages can be used to change patches in several linked 200-series pedals at once. The RV-200 is also compatible with the BOSS ES-8 and ES-5 switchers.

All the key parameters can be accessed directly using the front-panel rotary controls and/or the reverb algorithm selector dial. Essentials such as reverb time, effect level, pre-delay and high- and low- pass filters are right where you need them, along with button control for five reverb density options and a parameter knob that changes function depending on the current reverb type algorithm. For example, in SlowVerb mode, the parameter knob controls the dry sound level, whereas in Spring mode it sets the number of springs from one to three. The Time knob can also be pressed to turn it into a patch-selection control for the first four memories or to enter manual mode. Unlike the deep editing of the 500-series pedals, Parameter accesses just a single additional parameter relating to the current reverb type. Buttons and associated rows of LEDs can also be used to select one of the four presets or manual mode.

Like other 200-series pedals, the RV-200 supports various mono and stereo configurations.

Depending on how the footswitches are assigned, the user can bypass the effect, scroll through memories or access a Hold function to generate continuous reverb. Other performance options that can be controlled from the footswitches include Warp (simultaneously controlling feedback and volume), Twist (a type of spinning reverb sound with deep modulation), and Fade. Despite the sophistication of what goes on inside this pedal, the simplified parameter set makes it very easy to set up effects either from scratch or by tweaking existing presets. On a more complex pedal I would have bemoaned the lack of a software editor, but here, you really don't need one.

The quality of the conventional reverbs is excellent, and Shimmer and SlowVerb are great for smooth, ambient treatments too. The new Arpverb is very atmospheric, adding sequence-like effects to the tails. I'm less sure about the authenticity of the spring emulation, which seems to have quite an electronic edge to it, but it's a musically useful treatment all the same.

MIDI In and Out are on TRS minijacks, and allow for remote Program Change switching of multiple connected 200-series pedals.

There's also a very useful Delay effect which offers repeat echoes and reverb at the same time, in addition to Modulated reverb and that old favourite, Reverse reverb.

So there's a lot to like here, and having user memories means those using it for live performance will be freed from forever bending down to adjust knobs. If you want the full-fat experience, the RV-500 will still be the beast of choice, and that model also includes a Space Echo emulation that can be used alongside the reverb. But for easy access to all the live essentials, the RV-200 has it nailed. *Paul White*

$ $269.99.
W www.boss.info

Newfangled Audio Recirculate

Delay & Effects Plug-in

Newfangled Audio's Recirculate, which is sold through Eventide's online store, is a feedback delay/echo that can process the feedback signals in various ways to make the repeats change character, with the change becoming more pronounced each time the signal passes around the feedback loop. While old-school echo plug-ins might do something similar with saturation and EQ in the feedback loop to emulate tape echo, and maybe with some pitch variations to emulate worn mechanics, Recirculate is designed to go well beyond that and offers something genuinely different. That said, it can still do a pretty mean tape echo emulation if that's what you want. The delay times can be set independently for the left and right channels and there's also a crossfeed option.

Supporting all the common plug-in formats, including AAX, Recirculate is authorised using an iLok account, though you don't need a physical iLok dongle. It has a resizable window and a choice of light or dark colour schemes amongst other user settings. There are global level adjustments for the input and output as well as an internal auto-gain function that compensates for any level changes resulting from the Drive control.

Occupying the top half of the GUI is the Stem Display, which shows the delay taps and allows you to set the delay time and feedback. Delay times can be sync'ed to tempo or steps, and you can set separate

delay times for the first echo and for the repeats, again independently for the two channels. Above the Stems display are four settings sections. Grid is where you choose to work in Notes, Steps or milliseconds. Tempo selects host sync or free adjustment, while Snap allows the Stem display to snap to note divisions, dotted note divisions, triplet divisions or to turn snap off to facilitate manual adjustment, with delay times from 10ms to one second available. Behaviour selects from Standard, Pitch Warp, Granular or Ping Pong.

In Standard mode you'll find all the familiar clean delay treatments, The Soundstage section, at the bottom right of the GUI, allows you to add further processing in the form of Spread, Chorus and Reverb, all of which may be used simultaneously if you wish, and each has an adjustable depth and parameter setting. In Pitch Warp Mode, changing the delay time causes pitch changes, as with a tape echo, while the Soundstage section changes to offer Spread, Modulation and Reverb. Switch to Granular and the Soundstage section offers Spread, Spray and Reverb. Spread offsets the left and right delay times slightly, to produce a wider-sounding effect. Spray and its associated Size parameter set the grain size and randomisation. Finally there's a Ping Pong mode. This bounces the repeats from side to side, has a Chorus effect in the centre Soundstage slot, and replaces the Xfeed parameter, below Spread, with Pingpan, which sets how the first repeat is placed in the stereo field.

In the main display, stems can be dragged to set the time of the first delay, the repeat delays and, by dragging up and down on one of the repeat stems, the feedback amount. Alternatively these can be adjusted in the Delay Beats section. The delay settings can be independently adjusted left and right, or linked if you prefer. In the mixer section below, the left and right feedback can be adjusted as one or separately, and further faders set the wet and dry levels. Any effects used can also be added to the dry signal. Below the mixer is a dynamics section offering compression, ducking and transient shaping, the last of these attenuating the decay portion of the sound so that the transient dominates.

There's a choice of filter types too: resonant, notch, scoop, combo and slope. These can be selected independently for the high- and low-pass sections, and a small display shows the shape of the filter response. They work inside the delay's feedback loop so that subsequent repeats receive more filtering. A section called Character allows you to dial in some distortion and noise, again with a short menu of different types from which to choose.

Recirculate is an incredibly versatile delay, and while its granular facilities might not rival those of a dedicated granular delay, they make it possible to create delay treatments that range from gritty lo-fi to endearingly silky. It lacks the traditional sort of multi-tap facility that would allow you to emulate a tape echo with three or four heads, but setting different left and right delay times will get you close to that anyway. Having such a comprehensive filter section along with modulation, reverb, various flavours of saturation and a choice of 'dirt' means that you can go from conventional delay, via tape and analogue emulations and well into the world of sound design. As long as you don't go overboard with the settings, most of what you can dial in seems to be musically useful too, so in all this is a very flexible delay that could slot into just about any musical genre. It's good value, sounds good and has a very user-friendly interface. What's not to like? *Paul White*

$ $99 (discounted to $49 when going to press).
W www.eventideaudio.com
W www.newfangledaudio.com

Electro Harmonix Pico Attack Decay Envelope Pedal

For those who want to add a slow attack to guitar sounds, there is, of course, the option of doing so manually by manipulating the guitar volume control or using a volume pedal, but several decades ago Boss came up with the analogue Slow Gear pedal that did the job for you automatically — as long as you played single-note lines that would allow the analogue circuitry to identify individual notes. Move ahead a few decades, and Electro Harmonix took the idea further in their digital Attack Decay pedal, which we reviewed back in July 2019 (https://sosm.ag/ehx-attack-decay). Billed by EHX as a 'tape reverse simulator', not only can this do the slow-attack thing but it can also shorten the natural decay of the sound, and is able to cover everything from bowed guitar effects, via reverse-like envelopes, to turning a guitar into a faux banjo. Unlike the old Slow Gear and its many emulators, the Attack Decay also has a poly mode that can follow the individual notes in chord arpeggios and the like.

This new Pico version drops the second footswitch, the effects loop and the harmonic distortion capability of the original Attack Decay, but it retains all of the essential envelope controls, including a poly mode activated using a button at the top of the panel. The attack and decay times have separate controls for adjustment plus there's a control for overall volume and another for sensitivity, making it easy to match the triggering to your playing style. There's even a secondary blend function that gives you the option of mixing wet and dry signals for when you don't want to start a fade from complete silence. The pedal, which has a buffered bypass, comes with its own PSU and there's no battery power option.

» Like the original, the pedal identifies individual notes extremely well and it has plenty of attack range, from just softening the attack transient to a slow bowing type of fade-in. If you also want to make use of the decay control, then the standard mode of operation gives the most natural-sounding results. But if you're happy to leave the sound decaying naturally, poly mode is very effective at tracking arpeggios: rake across a chord, and each note will trigger its own attack — this is very impressive stuff! (The decay control does still work in poly mode, it's just that the results can become a bit unpredictable at shorter decay times.)

The Pico Attack Delay perhaps won't be considered a pedalboard essential by every player, but it has lots to offer the more experimentally minded — especially those who have an appetite for making electric guitars sound like something else. So it's also worth me highlighting that this pedal can work particularly well in tandem with others. For example, putting a shimmer reverb after the Pico Attack Decay produces a fairly convincing analogue string sound, or put an octave pedal up front and you have a pseudo-cello.

Of the more affordable envelope pedals out there, I think it's fair to say that this one probably has the most to offer in the way of flexibility, especially if you're interested in changing the decay time of the sound, as well as adding a slower attack. *Paul White*

$ $136.90.
W www.ehx.com

Minimal Audio Swarm Reverb

Reverb & Delay Plug-in

There are so many reverb plug-ins for Mac and Windows DAW software now that, in order to stand out from the crowd, a new one needs to offer something a little out of the ordinary. Supporting all the mainstream plug-in formats including AAX, Minimal Audio's Swarm Reverb does exactly this by combining a fairly conventional algorithmic reverb with modulation and an assertive ducker that, at extreme settings, can produce a slow fade-in of the reverb. A graphic of the early reflections is shown in the centre of the GUI, and the Size knob adjusts their spacing from short and tight to long and grainy. An Attack control adjusts the amplitude of the various reflections, giving a normal decay at its minimum setting, but a slow build-up at longer settings. When modulation is added, the delay timing of the early reflections moves in opposite directions in the left and right channels. The modulation range goes from a gentle slow chorus, via weird caves to a swarming hiss at faster rates and high depths. A mouse-over tooltip mode explains the controls, which is helpful.

There are level controls for both the input and output, as well as high- and low-pass input filters that affect only the wet sound. A separate low-cut diffusion filter with adjustable gain and frequency is applied to the early reflections diffusion generators. Most reverbs feature a pre-delay to offset the start of the reverb, and here we get a full 500ms of delay range, with adjustable feedback and a tempo-sync option, making this feature more like a conventional digital delay than a simple pre-delay.

For the late reflections or reverb tail, there are controls for Decay and Damping as well as a Balance control that adjusts

the mix of early reflections and tail. The ducker is adjustable for both rate and time, and it's very effective in killing the reverb until the input level drops, then allowing it to swell back in, creating a convincing bowing sound. At lower settings the ducked effect can really help to keep the sound clean, by gently reducing the amount of reverb being added when the dry level is high. Finally, there's an output limiter that can either be off, gentle or punchy. These limiter modes can be useful but can also sound a bit dirty on some material, in which case it may be best to bypass them. A horizontal fader sets the master wet/dry balance.

Despite there being only a single reverb algorithm, the modulation, delay section, ducking and attack parameters make it possible to conjure up a wide range of reverb treatments that go from vintage algorithmic spaces to otherworldly special effects. By comparison with reverbs that aim to sound very natural, Swarm Reverb is undeniably more coarse-sounding — but that's precisely what gives it such an endearing character. It probably wouldn't be my first choice for drums and percussion, but for other instruments and voices it can be supremely effective, especially if you want to stray outside the more obvious reverb treatments. *Paul White*

$ $49. Also included in the Minimal Audio All Access plan.
W www.minimal.audio

JMG Dropkick

Kick-drum Processing Plug-in

Available from United Plugins and authorised using a personal key file that permits use on multiple computers, DropKick is a comprehensive set of tools for reshaping kick drum sounds, but it manages this with just a handful of rotary controls. All the common plug-in formats for Mac and Windows operating systems are supported, and updates are free.

The large Warp knob in the Phase section uses all-pass filters to offset lower frequencies in time. The filters are analogue-modelled, so there are non-linearities that can add a little more warmth and change the character of the attack transient. A search button automatically homes in on the fundamental frequency of the kick sound being treated, and this usually works very well, but this can be set manually if you prefer. X Filter selects how many

all-pass filters are deployed. The more filters there are, the greater the phase shift will be, but you don't have to think too hard about this — just listen and trust your judgement. In and out level controls compensate for any level changes due to the processing.

In the Power section, which again has a bypass button, there are Freq

(frequency) and Reso (resonance) controls for a high-pass filter, which can be set to two- or four-pole, with 'fat' and 'normal' versions of both. Both the Phase and Power sections can be locked to prevent further changes to their frequency settings. A Trim control can be used to shorten beats, including those that have been extended by the resonant filter.

Fire is a psychoacoustic processor that resynthesizes and harmonically enhances the low end, the aim being to make the sound translate more effectively to smaller speaker systems. Another control, Over, is a compressor that tends to bring out high-end detail while controlling the mids and keeping the low end consistent. Meanwhile, Snap emphasises the attack transient and Clean is designed to tame boxiness and muddiness in the lower mids. Damp emulates physical damping by progressively reducing high frequencies using an envelope filter, without compromising the attack transient. There's also a limiter with Clean, Warm and Hot options, plus a Drive knob that adds up to 12dB of additional gain before the limiter. Meters show the in and out levels as well as gain reduction.

"There are tools to fit any style, from tight and snappy metal to EDM."

The range of control this plug-in gives you over the kick drum sound is impressively wide and there are tools to fit any style, from tight and snappy metal to EDM. If your kick needs more depth, more clarity, more tightness or a hefty resonant punch, it's all here, controlled by a very intuitive user interface. I observed a little graphical weirdness when opening the plug-in in Logic Pro on my Mac's second monitor, but other than that it worked just fine. There are the usual presets to get you going and tools for comparing A and B settings, but really you're unlikely to need this — it's just as easy to dial in your own settings and let your ears guide you. All in all, this is an impressive kick drum tool kit, and eminently usable. *Paul White*

$ $74. (Discounted to $20 when going to press.)
W https://unitedplugins.com

Instant Audio QuickMuse

Creative Randomisation Multi-effects Plug-in

With an interface that looks as though it were designed by an astrologer on magic mushrooms, Instant Audio's QuickMuse is a five-section multi-effects plug-in with a simple interface and some smart randomisation functions. Authorisation relies on a personal key code, meaning you can use the plug-in on all your own machines. There's a 15-day free trial, and I gather that updates are free. Internally, the plug-in uses up to 8x oversampling, at sample rates up to 192kHz and beyond

Rather than conventional names, the 64-bit effects are designated as elements: Fire, Air, Water, Earth and Aether. Each element is represented by a symbol that doubles as a mix fader, and this gets brighter as you add more of the effect. Each element can be bypassed individually, and can also be locked to prevent changes when randomisation is applied. At the bottom of the GUI are high- and low-pass filters and level controls for the input and output, plus gain and global wet/dry mix controls. A circular button in the centre of the screen invokes a global randomisation process for all the unlocked element effect settings and also determines their connection order. Smaller circular icons around it randomise the functions of individual elements. There are presets categorised as Bass, Guitar, Drums, Vocals, Synths and some for individual elements, though the real focus of this plug-in is its random paradigm.

Fire controls warmth and distortion, while water adds movement, drawing on tremolo, flanger, phaser, auto-pan, filter modulation and vibrato. Air, based on both reverb and delays, adds space while Earth employs chorus, stereo widening, multiband dynamics and shelf EQs, to draw out more of the character in a sound. Aether explores the realms of granular synthesis and spectral transformations, including multi-pitch shimmer reverb, reverse delay and granular delays.

Other than random selection, editing is limited to the dry/wet mix of each element, and changing the effect order by dragging the icons at the bottom of the GUI. The randomisation process is described as being 'smart', insomuch as it endeavours to select settings that will work well together. Even the bypass is smart enough to avoid clicks and glitches when bypassing the plug-in, and when the plug-in comes across a section of track with no audio, it sleeps to conserve CPU overhead, whichever plug-in format you're using.

QuickMuse's complete reliance on random selection may seem a bizarre concept, but in practice it works well, producing mainly gentle and textural, modulated-reverb type treatments. I found that many of the combinations worked well on ambient music parts and sometimes even on whole mixes, but there's also great potential for using QuickMuse on individual audio tracks, especially when you need to give something the soft-focus treatment. When you do come across an effect you like, you can to save it as a preset (the chances of getting back to it using randomisation are less than those of winning the lottery!) and gradually build a library of useful and interesting effects. Whether you're looking to create soundscapes or just after some inspiration when you run out of ideas, QuickMuse offers creative potential. *Paul White*

$ $52.
W https://unitedplugins.com

STEP UP YOUR SYNTH CHORDS

Polysynth Masterclass

Do you find yourself reverting to the same familiar chords? Technology can help...

ALEX BALL

One of the pitfalls of becoming familiar with an instrument is that we can default to the same kinds of shapes and patterns we've used over-and-over. When I was learning guitar I was shown a trick that helped break out of this, and a comparable idea can be employed with polyphonic synthesizers (or keyboard instruments in general).

With the guitar the trick was simple: the player simply tunes to a different tuning, such as open G or DADGAD. As re-tuning causes all of the notes to move from their normal positions on the fretboard, the shapes you would default to don't work, or if they do, they lead to a different combination of notes from usual. This re-tuning therefore forces the player to find different shapes and to use their ear and then, once back in good ol' EADGBE, to start thinking in a different way about approaching chord voicings in general.

The way I'm going to do this on a synthesizer is to choose an instrument that has at least two oscillators per voice. This can be hardware or software, it doesn't matter, but it's crucial that two pitches sound when each single key is pressed because of the dual oscillators.

If you don't want to use a synthesizer, you could load up two instances of a piano patch in your DAW, set them to the same MIDI track and then view them as 'oscillators' 1 and 2 when reading the explanation below.

As you'll see, if we tune the second oscillators at different intervals to the first and then play chords, all sorts

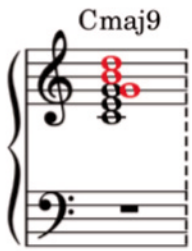
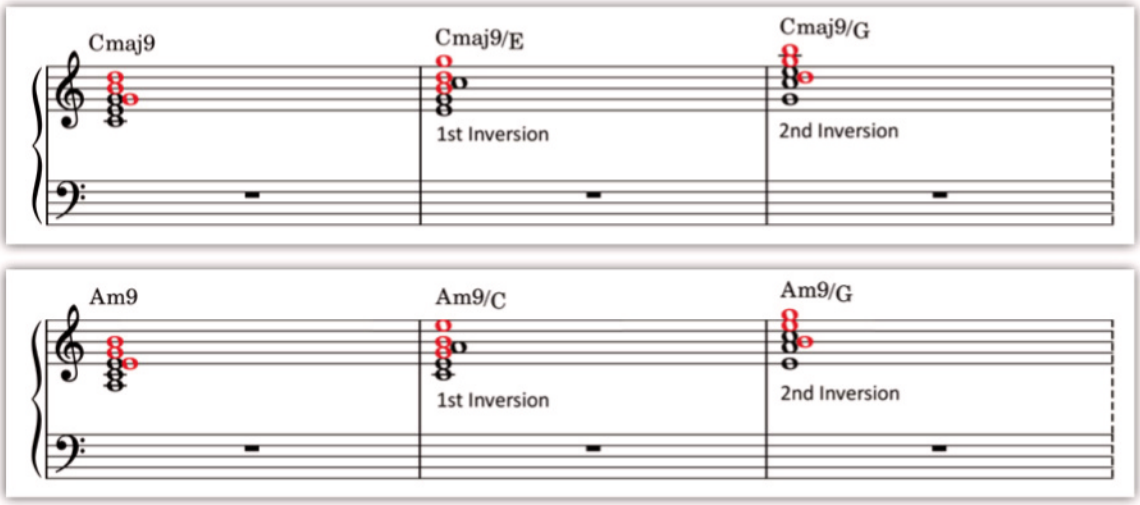



Figure 1: The notes physically played on the keyboard have black noteheads. These are also the pitches that sound from the first oscillators. The notes with red noteheads are those created by the second oscillators that are tuned at an interval — in this first case, a perfect fifth.

Figures 2 and 3.



of unexpected and interesting things happen, even if we play the same shapes we're familiar with. Better still, we can analyse what is happening and take ideas from it.

Tuning In Perfect Fifths

In this first example, we'll tune our second oscillators seven semitones higher than the first, so that the interval between them is a perfect fifth. If we play a simple C major triad, we immediately get something much more interesting. As you can see from Figure 1, our first oscillators give us the notes we played (C, E, G), but the second oscillators give us G, B and D at the same time. Putting the notes in order, we have C, E, Gx2, B, D, which is C major 9 (with two fifth degrees).

If we invert the original chord, we still get C major 9, but with some lovely voicings. As shown in the notation in Figure 2, playing the first inversion of our C major triad (E, G, C) gives us E, G, B, C, D, G and playing the second inversion (G, C, E) gives us G, C, D, E, G, B. Once we put these inversions into the context of a chord sequence, their usefulness will be revealed.

Next, let's take a look at the relative minor triad, A minor. If we play A, C, E then the first oscillators give us those notes, but the second oscillators give us E, G and B above them. Putting those notes in order, we have A, C, Ex2, G, B which is A minor 9. Again, inverting the basic triad gives us more interesting voicings of the chord, as in Figure 3.

But why stop there? We've only used three fingers so far, so what if we added a fourth? If we play a C add 2 chord (C, D, E, G) then our oscillators give us C, D, E, Gx2, A, B, D, which is C major 9 (add 13), as shown in Figure 4. That's impressive, considering we only added one extra note to a basic major triad!

This works for minor chords too. As you can see in Figure 5, if we play an A minor (add 4) chord (A, C, D, E), our oscillators give us A, C, D, Ex2, G, A, B, which is Am11.

Now, here comes the challenge. If we start playing chord progressions, as well as the juicy extended chords we've intentionally created, we'll get some nasty dissonant ones, so we'll have to use our ears and start moving notes around within our chords until we find something we like. This is, of course, the entire point of the exercise: we are trying to break out of the same old chord progressions and note combinations. Figure 6 shows an example chord progression to try with oscillators tuned in fifths.

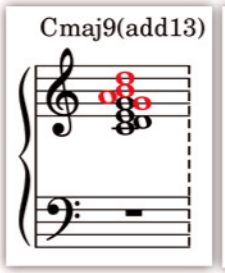


Figure 4.

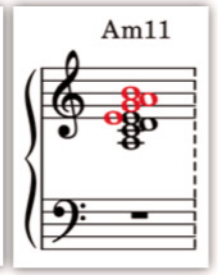
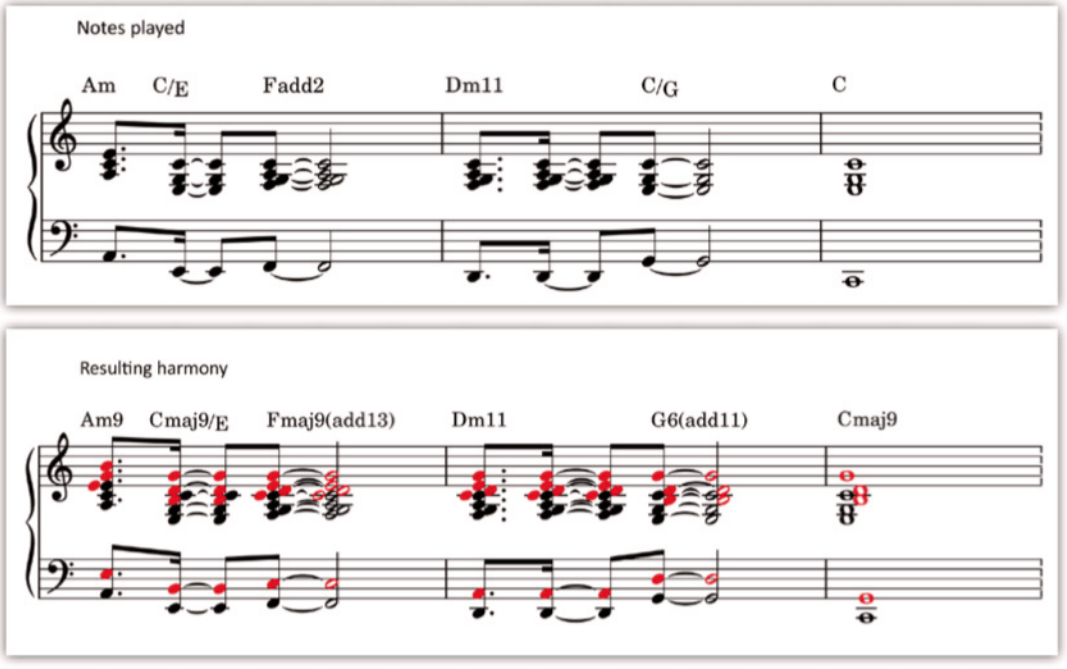


Figure 5.

Tuning In Perfect Fourths

Let's try the same principle, but with our second oscillators tuned in fourths. Now our basic C major triad gives us something different: C, E, F, G, A, C, to be precise, which isn't really a C major chord, but actually an inverted Fmaj9 chord instead. I did say that this will mix things up! A quick mental transposition tells us that playing the shape of a G major triad will give us a C chord. Again, we can add a fourth note, as we did before, to get some extra spice, as shown in Figure 7.

Figure 6.



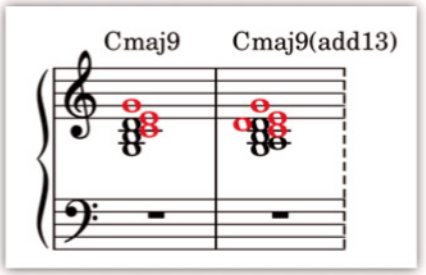


Figure 7.

Our A minor triad also gives us something different: A, C, D, E, F, A, which is actually Dm9/A. Another quick transposition tells us that playing the shape of an E minor triad will actually give us Am9/E. Staying with this 'E minor' triad, if we add another fourth note (E, G, A, B) then we get E, G, Ax2, B, C, D, E which is a nice, squashy Am11/E, as in Figure 8.

Now is a good moment to mention that the transposing/bi-tonal nature of this technique can cause problems, but this can be helped by playing root notes in the left hand. In order to keep this separate, we need to set up a keyboard »

Figure 8.

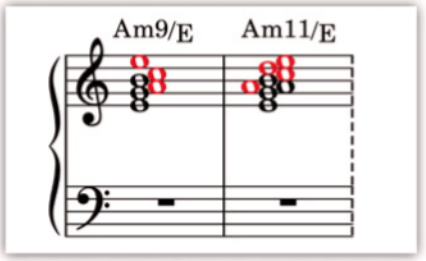
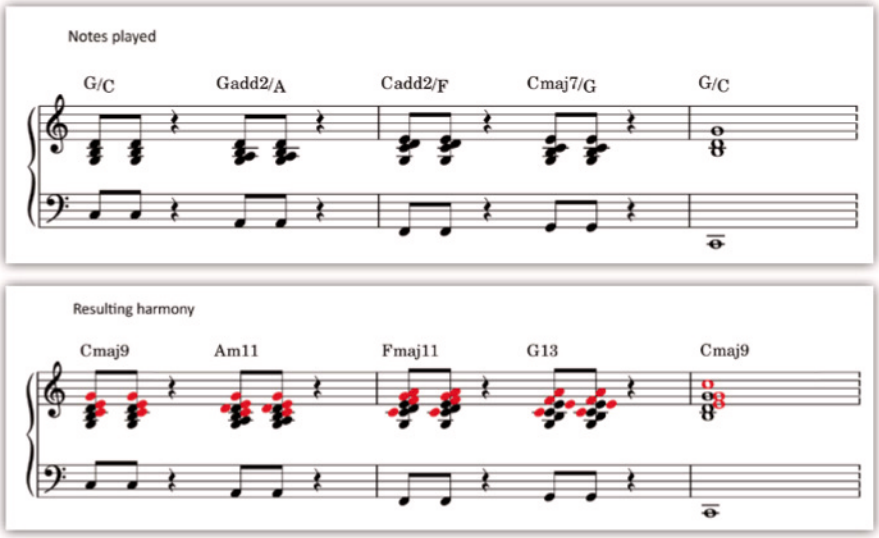



Figure 9.

» split where the left side of the keyboard has oscillators tuned in unison or octaves and not at another interval.

With this is mind, Figure 9 shows a chord progression to try with the right side of the keyboard, using oscillators tuned in perfects fourths, and the left hand in standard tuning.

Taking Things Further

And here's where things get even more interesting. What if we had two parts, one tuned in fourths and one tuned in fifths, happening at the same time? Now even the common I, V, VI, IV progression becomes a different prospect entirely. I'll stick to the key of C major and so our four basic chords would be C (I), G (V), Am (VI), F (IV). If we extend those chords to add the seventh degrees, then we can see the fuller context: Cmaj7, G7, Am7, Fmaj7. We need to be aware of this as we work out our voicings.

For the first synthesizer part, we'll use the perfect fifth tuning. There's an issue to work around, though, which is that the upper extensions of our chords (created by the second oscillators) give us some duff notes. For example, if we were to extend the G major that is V in the key of C major, this would normally be a dominant seventh chord, as mentioned above. With our oscillators in fifths, we get a major seventh chord (actually, major ninth) instead. To work around this, we'll need to try some different shapes and combinations of notes. F, G, D, E sounds right to my ears and, lo and behold, an analysis shows us that this gives F, G, C, Dx2, E, A, B. Restructure those notes (G, B, D, F, A, C, E) and we have all of the tones of of a G13 chord, which is an extended version of G7. Bingo!

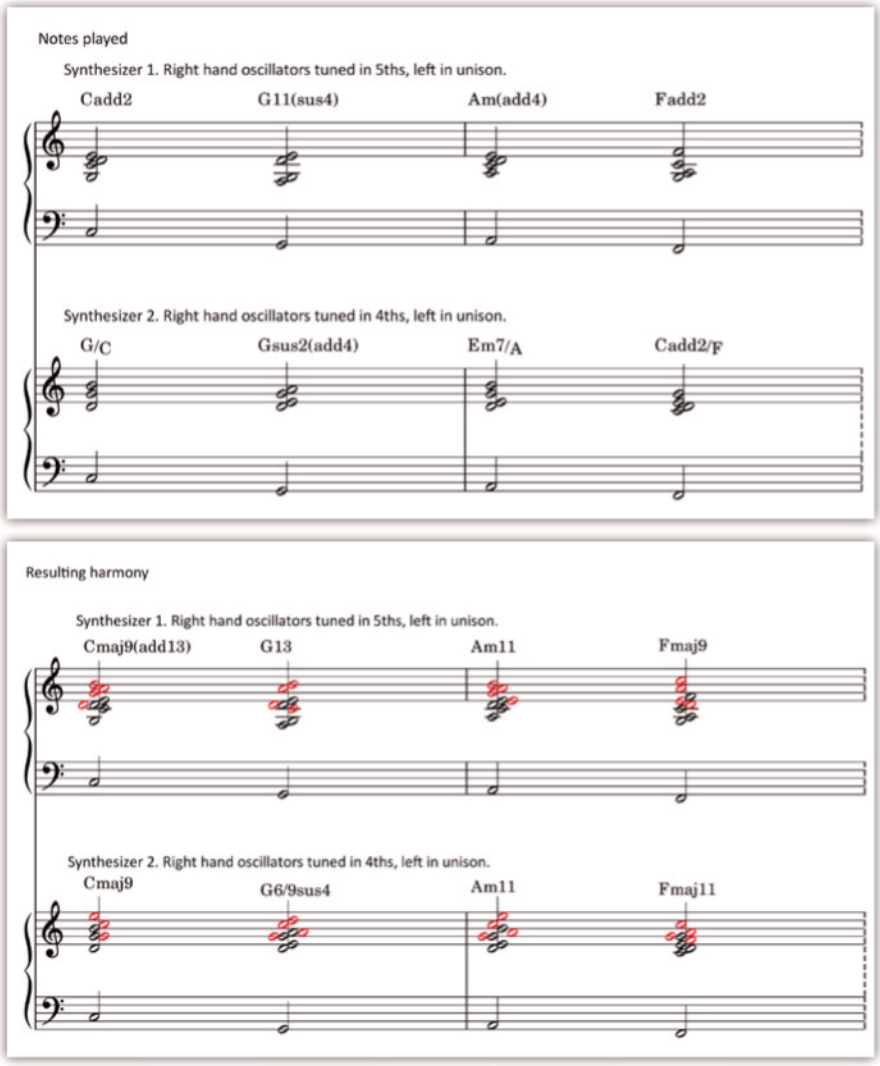
Over to the second part, and we're tuned in fourths. Again, to stay in the key we want, we need to play in, seemingly, a different key! We also still need to shift notes around to avoid certain bum notes being created by our second oscillators. For example, as with the first part, we wind up with major seventh chords (or extensions of) and there is a dominant seventh chord in our progression to consider.

Put these down with a unison bass beneath to tie them together and look at how dense our harmony has become in Figure 10, without us playing many keys.

So, what can we learn from this? This exercise gives us a glimpse at what extended versions of chords (and chord progressions) can sound like and so, if we tune our oscillators back into unison, we can look at bringing in some of those voicings and ideas in a 'normal' way by playing them on the keyboard. And then, just like that, we're no longer playing those tired old chords.

Audio examples for this article can be found at sosm.ag/synth-chords.

Figure 10.



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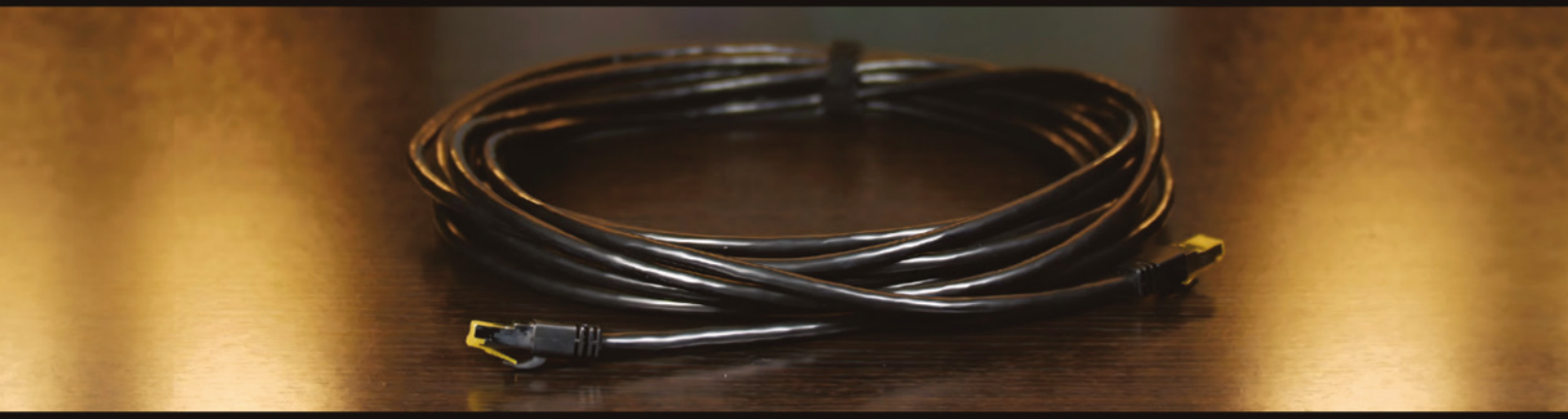

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On Specs

Why Specifications Matter: Part 4

We conclude our series with a look at how specifications can accommodate the foibles of human hearing.

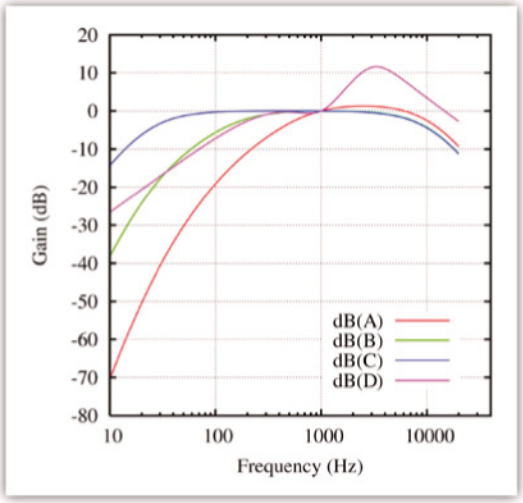
JOSHUA ISRAELSOHN

As mentioned in the first three instalments of this series, audio-equipment specifications can facilitate useful point-by-point paper comparisons, but these evaluations are often more informative when made in the context of your particular use cases, interests and preferences. Measurement systems, however, don't natively respond to audio signals the way human hearing perceives them.

The Weighting Game

Unless set to do otherwise, audio-measurement systems observe signals with equal sensitivity across the audio spectrum. This is appropriate for making good objective measurements but ignores the fact that a key element in most audio systems is one or more human ears, which do not exhibit such a flat frequency response. Human hearing is more sensitive to midrange frequencies — roughly 1 to 4 kHz — than to those at either the lower or upper ends of the audio spectrum. As a result, sounds at low frequencies and, to a somewhat lesser degree, high frequencies require greater intensity — sound pressure — to be perceived at the same loudness as a midrange reference tone such as the typical 1kHz.

To approximate the human hearing response at modest volumes, test systems use an A-weighting filter which rolls off both ends of the audio spectrum, thus approximating the frequency-dependent sensitivity of the human ear. The A-weighting filter resulted from data collected and published in 1933 by Fletcher and Munson that showed the non-linearly variable sensitivity of human hearing over both frequency and loudness. Their seminal work was refined in 1956 by Robinson and Dadson, which formed the basis of the ISO 226 standard equal loudness contours to which you may see references. The most recent revision of this was published in 2003.



The A-weighting curve is shown here in red, alongside other weighting curves that are used in some applications.

A-weighted measurements are used primarily and extensively for measuring environmental noise and noise in audio equipment. For example, you'll find an A-weighting filter (or an otherwise-named functional equivalent) as an option on most all loudness meters, such as those used in environmental studies and in specific industries where sound exposure is regulated.

In both loudness-metering and audio signal-processing applications, A-weighted measurements report numerically lower amplitudes (with units denoted dBA) than corresponding unweighted measurements in dB made with the same equipment under the same operating conditions. One might be tempted to criticise A-weighted measurements as putting marketing spin on the spec sheet to gain unearned advantage. But that misses the point of the measurement, which is to express a parameter quantitatively in a way that corresponds well with the qualitative experience of typical human hearing. Just be certain that, when comparing specifications for competing pieces of equipment, if one reports an A-weighted measurement for a specific parameter, the other does as well.

How Noisy Is My Mic?

Note to those disinclined toward maths: You'll never need to do any of the calculations shown below. They are given for spec-sheet insights and for the physics-curious. Read through and glean what you can if you like, but don't panic — there's no test and, in real-world practice, the maths are all done by equipment manufacturers' engineers to prepare their various spec sheets.

Among the common uses for A-weighted noise measurements are microphone specifications. Microphones come in two basic types: dynamic

and capacitor. Assessing the so-called self-noise of a dynamic microphone is easy: for all intents and purposes, there is none. To be more precise, the self-noise of a dynamic microphone is the thermal noise (also called the Johnson noise) of the microphone's output impedance, R, typically 150Ω. As with all noise measurements, the measurement bandwidth, Δf, is a factor — we'll call that 20kHz in round numbers. As the name 'thermal noise' suggests, temperature, T, is another factor, so let's use 20°C as typical, but expressed in Kelvin as the maths require, so 293K. Put it all together and you get the noise voltage

$$V_n = \sqrt{4kTR\Delta f}$$

where k is the Boltzmann constant. For our example, the self-noise of a dynamic microphone with 150Ω output impedance is 220nV, which is about -133dBV or -131dBu. (Recall, dBV is dB relative to a 1V reference; dBu is dB relative to 0.775V — a historically common reference level derived from 1mW into 600Ω.) This is at least a few decibels below the best modern preamplifiers' own electronic noise floors.

Capacitor microphones, by contrast, have integral powered electronics to convert the signals from their capacitive transducers into a useful voltage signal. They exhibit excellent sensitivity (output voltage for a given acoustic loudness) but the transducer's associated electronics necessarily include noise sources.

A convention common with electronic signal processors, particularly those with noise outputs that vary with settings, is to lump all noise sources that are distributed throughout the circuitry together and model them as a single noise source that appears at the input of a theoretical, functionally equivalent, noiseless apparatus. The quantity attributed to this modelled noise source is called the Equivalent Input Noise or EIN. This convention is often followed in specifying the self-noise of capacitor microphones, but due to microphone's key function, transduction from the acoustic to voltage domains, that puts the EIN in the acoustic realm as if it were a small acoustic noise-making device at the capsule location with a noise amplitude measured in dBA.

Recall that all measurements expressed in dB are relative to some reference quantity. In this case, it is the threshold of human hearing at 1kHz that defines 0dBA (equivalent to an acoustic pressure of 20 µPascal). Obviously, we can't measure that virtual noise source directly, but we can measure the physical noise output voltage of the capacitor microphone's electronics. So, let's say we effectively mute the capsule, measure the microphone's resulting output noise voltage (with A-weighting), and get a noise figure of -114dBV (A-wtd). We can then convert that into a raw rms voltage:

$$\text{Vnoise} = 10^{(-114/20)} \times 1^{*} = 0.002\text{mVrms}$$

(*The x 1 is because of the 1V reference in dBV. If the measurement had been dBu the multiplier would have been 0.775)

Let's say the microphone's sensitivity is a typical 20mV/Pa. We can then calculate the noise voltage as a decibel ratio with that 20mV reference:

$$\text{dB} = 20 \text{ Log } (0.002/20) = -80\text{dB}$$

The sensitivity reference is 1 Pascal, which equates to 94dB SPL, so the microphone's self-noise is 80dB lower than that:

$$94\text{-}80 = 14\text{dB SPL}$$

Therefore, the microphone's EIN figure is calculated to be 14dB SPL.

Like most analogue equipment, the signal-to-noise ratio (SNR) of microphones is calculated relative to a standard reference level — in this case, an acoustic input of 1 Pascal = 94dB SPL. Here the maths is trivial if the microphone's EIN is given in the spec sheet: SNR = 94dB - EIN. So, our previous example with an EIN of 14dBA results in an SNR of 80dB, which a pretty good spec. Similarly, if a microphone's spec sheet provides the SNR, you can calculate the EIN, if you have use for it, just by flipping the equation around: EIN = 94dB - SNR.

At some elevated acoustic pressure the microphone's electronics will eventually overload, so microphones' dynamic range is commonly larger than the SNR. For example, if our microphone has a maximum acoustic input of, say, 115dB SPL, the dynamic range will be 115-14 = 101dB, which is 21dB greater than its SNR spec suggests.

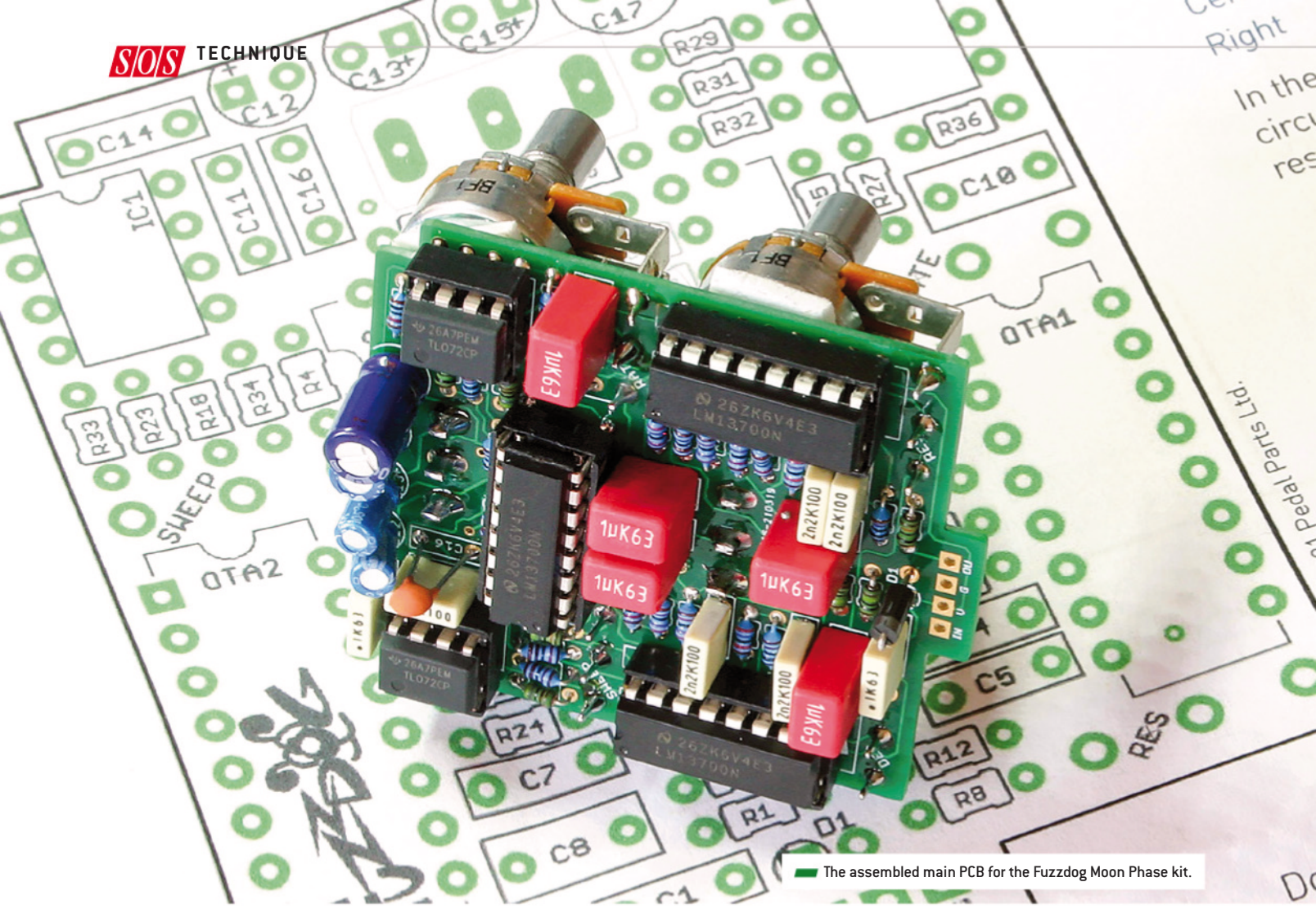
Lastly, remember that some of microphones' most important characteristics either don't appear or are only hinted at on microphone spec sheets. For example, frequency response curves can suggest that a certain microphone design is biased for vocal and midrange instrument applications. But two such microphones of different manufacturers may offer very different sonic nuances in how they replicate an instrument or voice. Similarly, a directional microphone's pattern often is a function of frequency, but how that affects the sonic quality for off-axis sound sources is difficult to read from pattern maps. So, while specs provide a useful way to narrow down options when comparing audio equipment, there are additional qualities, which are sometimes more subjective, that are worthy of your consideration.

How Much Do Specs Matter?

Comparing spec-sheet performance is one way to evaluate competing pieces of audio equipment but, as previously mentioned, it needs to be done in the context of your particular audio applications, needs and interests. For example, low-noise microphones, mic pres and audio interfaces are great if you're recording ambient microphones as part of your miking design to pick up the natural acoustic qualities of a performance space. But if you're a sax player in a funk band looking to put together an on-instrument rig, chances are noise performance isn't as high a priority as, say, build quality and physical robustness. Only your specific use cases can inform which equipment specifications are most important for a given equipment purchase.

General recordists, especially independents in the early stages of their career, may benefit from taking a note from the soundless world of still photography, where it is oft said, "The best camera in the world is the one you can put in your hands right now." The same notion applies to purchasing audio gear: You'll always have to work within a set of equipment constraints, but optimising your equipment selection process to get the most from your budget is more productive than holding out for the numerically best specified model you can't afford. Consider your understanding of spec sheets as one tool in your kit to help you do just that and get the best gear your money can buy.

This series was produced in association with Audio Precision, Inc.



The assembled main PCB for the Fuzzdog Moon Phase kit.

DIY Effects Pedals

Building Your Own Stompboxes

Making your own pedals can be a lot cheaper than buying them — and with many of the self-build kits now available, you won't even need a deep knowledge of electronics.

PAUL WHITE

With so many circuit schematics available online, and numerous companies offering DIY guitar pedals in kit form, there's never been a better time to build your own stompboxes. Not only can this save you money, but you can also recreate coveted pedals that are no longer in production. You can also adapt them, for example by combining multiple effects in one bigger enclosure, or building them with a modular synth system in mind.

But what are the realities of rolling your own? In this article, I'll explain the skills and tools you need, and discuss a recent kit-based project, to give you a better feel for how it works in practice.

Tools, Parts & Tactics

There's some initial outlay for tools, of course, and if you don't have most of them already, you're unlikely to enjoy any cost saving unless you plan on making at least a few kits (or need the tools for other jobs). You'll need a decent 30-50W soldering iron, ideally one that's thermostatically controlled and has a fine conical tip. You'll also need a multimeter. These days, meters that can read capacitor values are very affordable, and as some capacitor markings can be a bit ambiguous, I'd recommend spending just that little bit more to get one. You'll also need some small wire cutters and a 'solder sucker' desoldering tool, for those occasions when the solder ends up where it shouldn't. Small pliers are useful for holding awkward components and as a heat sink when soldering. I also find a bright work light and a desk magnifier very helpful.

When assembling the final pedal, as well as screwdrivers you'll also need spanners (or a small adjustable spanner) to tighten the nuts around the pots, footswitch and sockets. If you plan to drill your own case, you'll also need an electric drill and a some suitable bits. The standard-size compact pedal case that can take a battery is designated 1590B. Mini pedals that don't have space for a battery usually reside in the narrower 1590A case.

As long as you can follow instructions you don't need any deep electronics knowledge, but you do need the ability to solder tidily. When you're soldering semiconductors and chips, you need to do it quickly, too! This isn't a soldering lesson (there are lots of good ones on YouTube) but the key thing to remember is to heat the joint first and then feed solder to it, rather than carrying solder to the joint on the iron's tip. If you're new to this, don't start with anything too complex. Begin by tinning wires and seeing if you can make a tidy job of wiring an XLR or jack plug. If you can do that, you're probably ready for a pedal project. From there, you can hone your skills on a simple kit, such as a buffer or basic fuzz.

Some kit suppliers offer ready-drilled boxes at a slightly higher cost, but drilling your own enclosures isn't too hard. Generally, for the standard size die-cast zinc-alloy boxes, the 'proper' method is to use a bench drill press, but they're pretty easy to drill with a handheld electric drill if you secure the box in a suitable vice or adjustable work bench. My approach is usually to mark out the hole positions in pencil (most kit builds tell you where to put the holes), then use a centre punch to make a small dent in the case in the centre of each marked hole, to help prevent the drill bit from wandering. If you start by putting a smaller drill bit through all the marked holes first, larger bits will be less liable to stray off course, and if you don't have a bit that's quite large enough, you can always use a round file to enlarge holes.

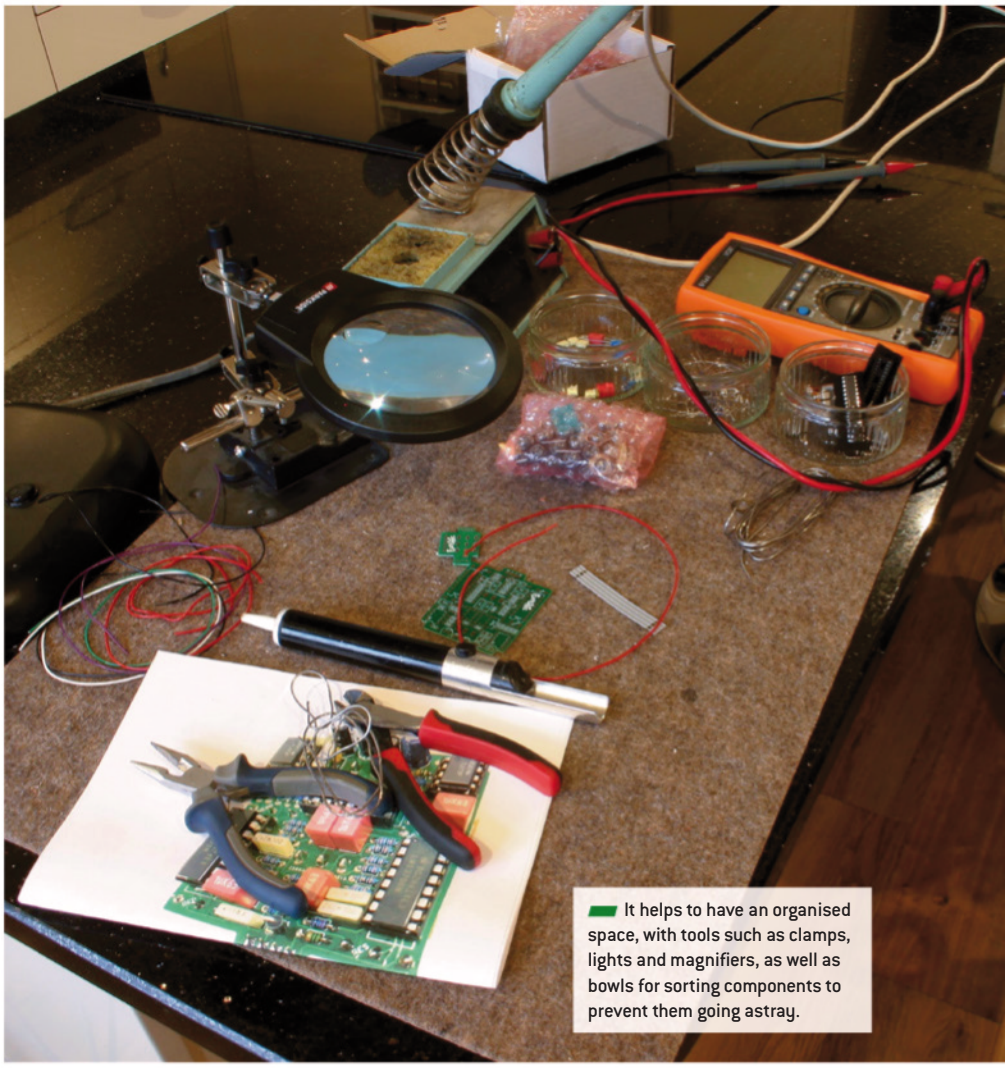
Inexperienced kit builders often worry about their ability to decipher resistor colour codes. Plenty of online charts explain how those coloured bands work, including the FuzzDog's free General Building Guide PDF (http://pedalparts.co.uk/docs/GeneralBuildGuide.pdf). But even though I trained in electronics, I still find some resistors hard to read where the colours aren't well-defined, particularly the tiny eighth-Watt resistors used in some kits. Their colour bands remain inscrutable even under a magnifying glass! So, I double check every resistor with my multimeter before inserting it into the PCB. It's a small time investment that can save a lot of time unpicking problems later on.

If you're more experienced in tinkering with electronics, you might be happy referring to online schematics and building your projects from scratch, using strip-board such as Veroboard, but buying the correct PCB makes for a tidier and more compact job, as well as being easier. You can usually track down the necessary components online, but note that buying these singly is much more costly than buying in bulk. If you plan on several builds, then it's at least worth investing in a pack of resistors that cover all the key values.

I suspect, though, that for their first build most would rather buy a full kit of parts, complete with circuit board and assembly instructions. There's also a safety net here, in that some DIY pedal companies (including FuzzDog here in the UK) offer a fault-finding service should you really come unstuck. There are lots of companies around the world who offer such kits, and personally, I've bought them from FuzzDog and JedsPeds. These UK-based companies offer kit options ranging from the bare PCB up to a full kit with a drilled enclosure, and can even build kits for you at a price (but where's the fun in that?!). I've built compressors, modulation pedals and distortion and fuzzboxes, and they've all worked well.

Orientation

The build I'm using to illustrate this article is FuzzDog's The Moon Phase phaser, which they rate as 'intermediate' in difficulty. It uses the aforementioned tiny eighth-Watt resistors, but unlike some phasers that require matched FETs, this one uses LM13700 transconductance amplifier chips, so there's no bias twiddling involved. As well as providing a number of kit options, Fuzzdog offer a small, optional daughterboard PCB that works for the majority of their builds and simplifies assembly. This fits on the back of the footswitch and can be cabled to the main board using a short length of ribbon cable (or separate wires). It also holds the status LED and terminations for the in and out jack leads and power »



It helps to have an organised space, with tools such as clamps, lights and magnifiers, as well as bowls for sorting components to prevent them going astray.

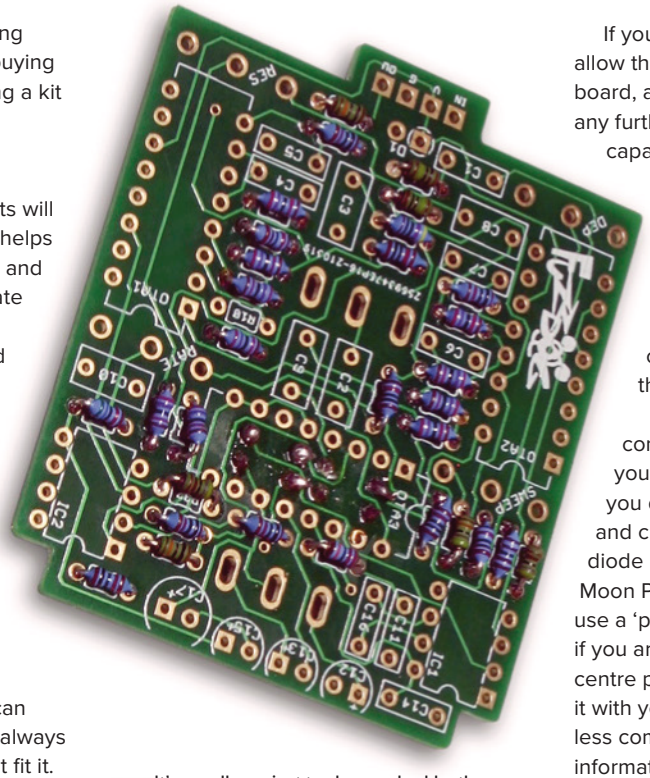
» connector, and helps keep the wiring inside the box nice and tidy. So if buying from them, I'd recommend choosing a kit option that includes it.

Setting up a clear workspace is essential, and having a few small containers on hand to hold the parts will help you keep things organised. It helps to sort out the resistors, capacitors and semiconductor devices into separate containers before you start. If you can fix up a way to hold your board secure while working on it, that can help too — you can buy some pretty sophisticated clamps, but for most pedal builds a couple of crocodile clips nailed to a block of wood works well enough!

When you unpack the full kit, you'll find a bag with all the components plus the necessary PCBs, pots, switches and connectors. Battery clips are normally provided for pedals that can accommodate a battery, but if you always use a pedalboard PSU, you needn't fit it. If you do plan to use a battery, check that your placement of switches and sockets still leaves room for the battery, and try to leave space for a piece of foam too — this can prevent the battery coming into contact with the circuitry.

Like many kits, this one has component numbers printed on the PCB, and a set of downloadable instructions tells you which values correspond to which part number. You can freely download the circuit diagram for any pedal FuzzDog offer, so you can easily check the complexity before you buy.

While resistors and small-value non-electrolytic capacitors can work placed either way around, electrolytic capacitors, diodes and many other components have to be oriented correctly. If a component has one leg longer than the other, the positive leg will be the longer one. Printing on the board indicates which end the 'stripe' on a diode should be, and which way around electrolytic capacitors should be mounted — their negative side is generally denoted by a minus sign on the capacitor body. Kits usually include sockets for the ICs (I'd recommend using these rather than soldering chips directly to the board, as changing a defective or blown chip that's been soldered to the board is a bit of a nightmare); both chips and sockets will have a small recess moulded at one end. Orientation is indicated both on the PCB and in a photograph, but if in doubt, there are also clear photos of a finished board. As long as you double check every component value and part number before fitting it, you really shouldn't run into trouble.



It's usually easiest to place and solder the resistors first. While it can take time, it's a good idea to check them with a multimeter before fitting.

On With The Build

I fit the resistors first, checking each one using the meter (set to measure resistance), to make sure the right resistors go into the correct places. Take extra care when measuring high-value resistors, because the resistance of your fingers can make the value look lower than it really is if you're touching both ends. Next, place the capacitors and, finally, any diodes, transistors, FETs and IC sockets.

The resistors in this kit are the small eighth-Watt types, but you could buy the larger standard quarter-Watt resistors and mount these vertically on this PCB if you prefer. Something to watch out for with this kit is that a small number of resistors must be wired on the rear side of the PCB before fitting the IC sockets, as their soldered joints end up beneath those sockets. These are clearly marked in the instructions and in the on-board printing, but it's well worth checking for such things before you dive in.

If your kit uses transistors, you can allow these to stand slightly proud of the board, as long as they don't protrude any further than the vertical electrolytic capacitors. Otherwise, they might foul the case lid that forms the base of the pedal. Crocodile clips or small pliers can be used as heat sinks on transistor legs and on diodes, though if you have a decent soldering iron and can make a good joint quickly, they are not always necessary.

When wiring the power connector, it's vitally important that you don't get the wires reversed. If you do, you could kill the transistors and chips, unless there's a protection diode in the circuit (sensibly, in the Moon Phase kit there is). Most pedals use a 'pin negative' wiring scheme, so if you aren't sure which tag goes to the centre pin of the power connector, check it with your meter. If you have one of the less common 'centre positive kits', that information will be included. If you plan to mix centre-negative and centre-positive pedals on the same pedalboard supply, you'll need a supply with fully isolated outputs and 'polarity swap' power cables to feed your centre-positive pedals.

For the flexible wires that are supplied with the kit, it pays to strip the ends, twist the individual strands and then tin the twisted end with some fresh solder. This will make it behave like a single piece of wire, which is easier to work with. And if you trim this tinned end at an angle, the wire is less likely to put up a fight when you try to poke it through a hole in the PCB or socket tag. When attaching wires to jack sockets or pots you can treat the ends in the same way, but I like to leave enough tinned wire exposed so that I can poke it through the hole in the tag and then wrap it around the tag with a small pair of pliers before soldering it. This habit, formed long ago during my 'ministry' electronics apprenticeship, keeps the wire in position as you solder it, which is not always so easy if you just poke the wire through the hole and hope it stays put while you solder. It also ensures that there's no protruding end to short out on something.

If the hole in a component tag is too small for the wire, tin both the wire end and the tag, then lay the solder alongside the tag, and heat while applying just a little fresh solder until you see a smooth joint. It can be tempting to blow on »

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» a soldered joint to cool it but you really shouldn't, as it may crystallise — either let it cool naturally, or use your pliers as a heat sink.

Sometimes, you'll find that pots need to be mounted directly on the circuit board. In such cases, it's important that you place an insulator between the back of the pot and the PCB, as this will prevent any short circuits. A thin piece of plastic rescued from a blister pack or even a couple of layers of duct tape will usually be enough to do the trick.

Other pots and switches need to be mounted on the case. If you've already drilled your case and the pots solder directly to the circuit board, a useful tip is to secure the pots and switches by just a single soldered lug, then do up the nuts finger tight. Once properly positioned in the case, you can then solder the other lugs knowing that they're in the right position, and that you aren't applying undue mechanical stress. As I mentioned, you have the option of enlarging the holes slightly with a round file. The washers provided with pots and switches should cover any unsightly gaps.

When you come to mount the finished circuit board in the case, adjust the nuts around the pots, switches and footswitch, so that these components don't stick out too far. But at the same time, make sure that they're not at the other extreme, where you can't get the back of the case on!

Problems & Testing

Common problems for novice builders include accidental solder bridges between lands or tracks on the PCB, and dry joints. If you manage to get solder where it shouldn't be, you can first try heating the offending blob, then tapping the board sharply on your work surface to see if that dislodges it. If that doesn't work, try the solder sucker. An alternative is to buy solder wick, which is a bit like coax braid — this draws up unwanted molten solder by capillary action, rather than suction.

> "As long as you double check every component value and part number before fitting it, you really shouldn't run into trouble."

Dry joints are another common issue. These occur when solder fails to bond with one or both of the surfaces to be joined. But as long as you keep your soldering iron tip clean by wiping it on the damp sponge that comes with it, you should be able to make clean joints — as I said, just remember to heat the joint first and then feed the solder onto the joint. If the solder flows around the protruding component lead, bonding to it, and then flows outwards to form a smooth mound on the PCB land around it, the chances are that you have made a good joint. My preference is to trim the component leads after bending them just enough to stop the component falling out of the board when inverted, but before soldering. You can then solder them without excess wire getting in your way. This also means 'stray ends' can't short out to nearby joints.

It's recommended to test pedal projects before fitting them in the case, and you can usually do this without worrying about having to resolder any wires as, invariably, the pots, switches and sockets (other than the power connector) fit into the case from the inside. Ideally, you'd test the board before wiring it to the footswitch too, and if you plan on making more than one pedal, something like FuzzDog's Simple Tester kit can be invaluable. This houses a footswitch, jack sockets, power connector and terminals for wiring to the main board, allowing you to check the board is working properly before you commit to wiring it up to its own footswitch and jacks. Only four connecting wires are required between main PCB and tester. If you run into problems once the footswitch is connected, you know your switch wiring is the cause.

If there's no signal through the effect when turned on (and any volume pot in the pedal is turned up!), start by checking that power is getting to the board. If the power is OK, use a magnifying glass to examine the solder side of the PCB very carefully, checking for dry joints or solder bridges. Also, confirm that ICs and transistors have been placed the right way round.

If you do get sound but the effect sounds wrong, check that you have put the right resistors and capacitors in the right places, and that any electrolytic capacitors have been installed the correct way around. If you use a wrong value somewhere, you might still get sound but it may be distorted, faint or just plain wrong. In my experience, a careful visual examination usually reveals any problem, as component failures are fairly rare (unless you really overcook semiconductors when soldering them in place). That said, if you've had to unsolder a wrongly placed component, it's just possible that you've damaged a track on the PCB, as they're very narrow. If so, you should just replace that damaged section with a physical wire.

If the pedal works OK until you screw on the case lid (ie. the base for your pedal), the chances are that something is shorting out on the lid. I always line the inside of my lids with a piece of duct tape to prevent that problem, but you also need to do a visual check to ensure no components are protruding that might foul the lid. Once everything is working,

The assembled electronics inside the supplied enclosure.

The finished article — while you *can* go to town with decoration, a label maker and a Sharpie can do what's needed!

you just have to decide on how to label the controls: a simple Sharpie or label printer does the job, but you can create your own custom style using waterslide decal paper transfers or paint if you wish.

Looking Back

So how was my Moon Phase kit to put together? While my build wasn't without a few head-scratching moments, none of that could be blamed on the documentation, which is creditably clear. Fitting all those small resistors took quite a long time because I checked each value with my meter, fitted the resistor, trimmed the leads and then soldered it in place before moving on to the next one. But it might have taken longer had I rushed it! I found the desk magnifier pretty much essential for this build, and the Simple Tester kit gave me an easy way to check the pedal was working before connecting the footswitch. You could manage without that, but that would mean soldering sockets to wires and reconnecting them once you'd fitted the switch. If you plan to build more than one pedal it'd be a worthwhile investment. Based on this experience, I'd suggest that beginners pick a kit with fewer components and perhaps larger resistors. But once you get the hang of soldering on small PCBs there's really no pedal kit you can't tackle.

Thanks to FuzzDog for providing the kit used to illustrate this article.

W FuzzDog: https://shop.pedalparts.co.uk

W JedsPeds: www.jedspeds.co.uk

Who says you can't create beats in Pro Tools?

JULIAN RODGERS

In music production circles, Pro Tools is often seen as the DAW of choice for people working in what might be called the traditional recording studio idiom, with the DAW being essentially a modern replacement for an analogue console and tape machine. While it's clear that DAWs are far more than capable of assuming the role of a virtual multitrack and mixer, it's also true that different DAWs enjoy followings with different users. In the same way that, broadly speaking, many composers favour Logic Pro, electronic music producers gravitate towards Ableton and people working in game audio like Reaper, Pro Tools has traditionally been seen by many as the first choice for people tracking bands.

DAW choice has always been influenced by perception as much as by features, with second-hand opinions being repeated online, often long after the missing features that inspired them have been addressed. Avid have been making a sustained effort to change the perception that Pro Tools is great for band recordings and singer-songwriters but producers and beatmakers would be best served elsewhere.

One widespread way of working is for a composition to start life in Logic or Ableton before being migrated to Pro Tools for vocal production and mixing — but the idea that one single DAW can't do it all is beginning to seem old-fashioned. Rather than the finishing steps being completed in a different DAW, why shouldn't the beat be created and completed all in Pro Tools?

An important part of changing the perception of Pro Tools among producers and beatmakers is to get them to try Pro Tools in the first place. This is why the introduction of Pro Tools Intro, a feature-limited but upwards-compatible and free version of Pro Tools was so important. Pro Tools First, introduced in 2017 and discontinued in 2022, wasn't attractive to potential new users. Pro Tools Intro, introduced later in 2022, definitely is.

A second step in attracting producers and beatmakers to Pro Tools is to offer the kind of creative environment that encourages experimentation that producers are accustomed to elsewhere. The non-linear composition environment pioneered by Ableton's Grid view has been co-opted by Apple in the form of Logic's Live Loops, and Pro Tools' new Sketch feature offers something similar. A free iPad app mirrors the functionality of the Sketch window in Pro Tools, facilitating easy collaboration. Another feature, introduced in 2020 and which supports this creative phase of the production process, is Ableton Link, which lets you synchronise musical beat, tempo, and start/stop commands across multiple Link-enabled applications.

The Separate Clip at Transients command provides an easy way to slice beats.

Collaborative jamming between iOS devices is a long way from tracking a rock band but if you need that, Pro Tools supports it. Unfortunately there is no recipe for creating a killer beat, but as Picasso said, "inspiration exists, but it has to find you working," and tools that facilitate the free flow of ideas put you in the best possible place to receive that elusive inspiration. So what Pro Tools-specific techniques are there beyond the aforementioned Pro Tools Sketch that can assist beatmakers? And if Pro Tools is so good at editing audio, how can that help? One such area is in the creation and manipulation of loops.

Loops & Breaks

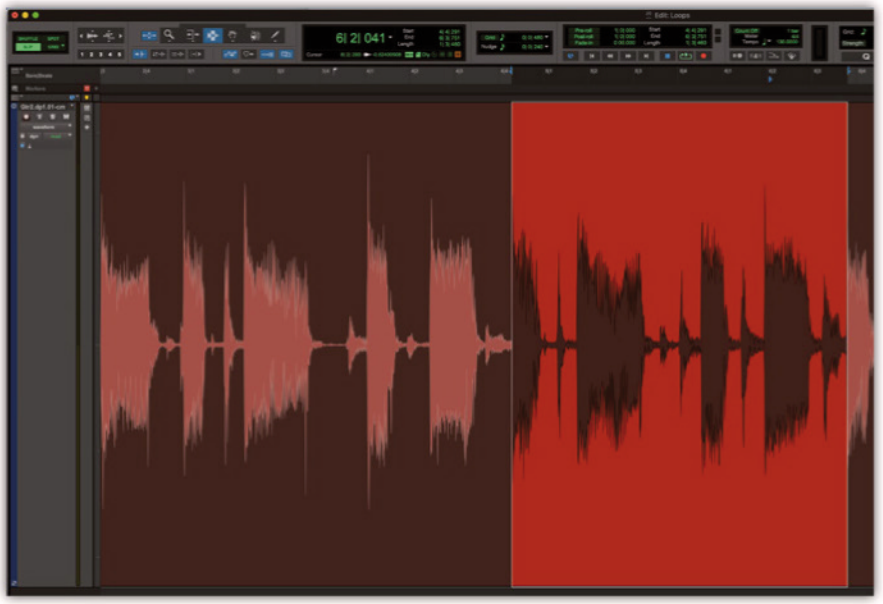
Breakbeats, originally samples of exposed drumming from '70s funk and the like, are central to many styles of production and while entire genres have built up around manipulation of well known breaks (think 'Funky Drummer' or the Amen break), the Wild West days of consequence-free sampling are behind us. But if you would rather not rely on third-party content, creating your own can still be inspirational, and in Pro Tools we have the ideal platform for detailed audio editing and sample manipulation. Particularly as, unlike many DAW's that use a separate sample editor, in Pro Tools sample editing happens in the familiar Edit page. Once a suitable piece of (legal) audio has been identified on an

audio track, one of the most powerful tools available to us is Tab to Transients.

Tab To Transients changes the function of the Tab key from advancing the Insertion Point along the timeline from one clip boundary to the next into a key that advances between audio transients within audio clips. The threshold for this isn't editable but the default works so well you might never notice. Adding Shift allows the creation of an Edit Selection and using these two keys enables the creation of fast and accurate loops. Once a loop selection is made, using the Trim command separates the loop and deletes the rest of the audio in a single operation (Command+T/Control+T). Useful if you don't need the rest of the audio.

Loops are fundamental to many styles of production and while using third-party loops is standard practice, creating your own feels far more authentic. But looping needs to be right. Making sure that the loop point doesn't contain any sustained elements which will cause obvious discontinuities is important — crash cymbals are a typical culprit. Checking the loop point can be as simple as selecting the Loop Playback mode by right-clicking the Play button or using the shortcut Command+Shift+L and playing through the entire loop, but a really useful function to try out if you work with loops a lot is Dynamic Transport. Select it from the Options menu or use the shortcut Command+Control+P. One of the advantages of Dynamic Transport for this kind of work is the blue triangle Playback Start marker in the timeline ruler. When selecting audio for a loop this can be placed just before the end of the Edit Selection to quickly audition the loop point without having to listen from the beginning.

Because audio clips are simply references to their parent audio files on disk, you can use the Trim tool to adjust the loop in and out points. Experimenting with loops can result in unexpected results and fun can be had by using Control (Start on a PC) with the standard +/- Nudge commands to nudge the clip's content rather than the boundaries. This way you can experiment with offsets between layers of loops. This reminds me of the time a friend misinterpreted where the downbeat was in a loop and created a brilliant groove as a result. Using this technique can be used to create melodic offsets, or on duplicated layers of loops with different processing applied to each, which can make for interesting delay-type effects.



The Dynamic Transport function in action.

Getting Creative

For instant unpredictability, something that was easy to stumble upon using hardware samplers but is more difficult to find with the easy GUI's of DAWs, try using the Separate Clip at Transients command from the Edit menu. This command is an easy way to divide up loops into their constituent hits and on a tick-based track a workflow similar to that found in REX files is possible. This method of manipulating the tempo of loops has been largely forgotten since the advent of good-quality DAW-based time-streching, but it has a definite sound. Something I like to do to change things up when a bit of unpredictability is called for is to Separate at Transients and then go into Shuffle mode and to 'stir' the contents of a loop. Like anything unpredictable the results vary, but because you can reverse out of poor decisions using undo, and snap hits to specific locations in Grid Mode, the results can be anything from inspirational to intolerable, just like my modular patches! When loops have been fragmented and reassembled the results can need some smoothing out, and the Batch Fades command can be used to quickly remove the inevitable clicks and pops.

If you're going for a more retro vibe GrooveCell might be just the ticket.

If this kind of mangling is too much for your needs, how about using the excellent GrooveCell instrument, which is available to 'on plan' Pro Tools users, to inject some old school charm into the proceedings? If you want to get the timbre but not necessarily the rhythm of a loop then you can disassemble it into its constituent hits using Separate at Transients, naming and exporting the clips as audio files by right-clicking on them in the Clips sidebar, and then dragging and dropping them into GrooveCell. Create your beat using standard MIDI features and whatever MIDI controller you have. If you don't own something with pads a keyboard is fine, and if you don't have a keyboard, hit Shift+K to open the MIDI keyboard window in Pro Tools. While you're in GrooveCell don't overlook the pitch controls, and for old-school flavour try the 'Old Sampler' or 'Drum Machine 1200' Drum Modes in Tab 2, or try Tab 3 for drive options from 'Analog Warmth' to 'Burning Gear'!

There's much more to the humble loop than just a drum break, but it's a great place to start to get properly acquainted with the ways Pro Tools can help a loop-based workflow. Once your loops are created there's the matter of how Pro Tools can help the quick creation of an arrangement, but that's probably one for another article.

Get better results from Iconica Sketch's strings with the Cubase Logical Editor.

Screen 1: The Iconica Sketch patch and MIDI track layout required for the workflow described here is very straightforward. Based on a simple chord sequence, some 'before and after' example MIDI clips are shown on the right.

JOHN WALDEN

Iconica Sketch is an excellent, compact orchestral library (it's under 5GB) included in the Pro, Artist and Elements versions of Cubase 13. As the name suggests, it's pretty good for sketching out ideas, but the sounds themselves are capable of much more than this — they come from Steinberg's full (190GB) and very impressive Iconica. Sketch includes multi-articulation patches for each major instrument section, but there are no 'full' ensemble patches. This is perhaps not such a bad thing, because although ensemble patches can provide instant gratification from a few simple chords, they're of little use if you're trying to write something a real orchestral string section might play — to do that, you really need to create separate lines for each string sub-section. To turn your simple chords into those individual MIDI lines for Iconica Sketch, Pro and Artist users need to do just a little patch configuration, and then take a quick dip in Cubase's Logical Editor. Although we're looking at strings here, the process can easily be adapted for the brass and woodwind sections.

Patch Work

The workflow I propose comprises two main steps, but there's an optional third one too. For the first step, the basic configuration is shown in Screen 1. This includes arranging the instrument patches in a single instance of HALion Sonic: these run from Violin I on MIDI channel 1 through to Basses on MIDI channel 5. Note that I've also edited the keyswitch assignments for the individual patches to span C-1 to F#-1. As I'll explain shortly, this will allow you to include real-time articulation switching for all the sub-sections. It's worth mentioning that all these patches offer the same seven performance articulations. For the shorter articulations, velocity controls dynamics, while for the sustained articulations you use the mod wheel to add crescendo/decrescendo to the performance.

Finally, alongside the Instrument track that's hosting HALion Sonic, I've added five suitably named MIDI tracks to the project. In their individual Inspector panels, I have set all of these to send MIDI data to HALion Sonic, but note that each uses the MIDI channel number required to target the desired instrument patch (channel 1 for Violins I, channel 2 for Violins II, and so on).

Now, select these five MIDI tracks (to record-arm them all) and play some four-note chords on a MIDI keyboard, plus any keyswitches needed to switch articulations in your performance. These full chords will get transmitted to all five tracks, and if you engage record when doing this, you'll end up with five identical MIDI clips. This might initially sound fairly epic — there are a lot of strings, so it makes a big sound — but it's not really what we want here, because every note in the chord is being played by all five sections. In a real string section, when it comes to basic chordal parts each sub-section usually tends to play a single note of the chord (unless a sub-section is playing 'divisi' and being divided into further sub-groups).

Think Logically

So, how do you thin out the five identical MIDI clips such that each one contains one particular note from the full chord and, when combined, the section as a whole plays the full chord, as might be the case with a real string section? You could edit each clip manually, of course, muting or deleting notes from each one to leave each sub-section playing a single note at a time. But that task gets tedious very quickly, and can interrupt your creative flow. Thankfully, with just a little work in the Logical Editor, you can automate this process pretty easily.

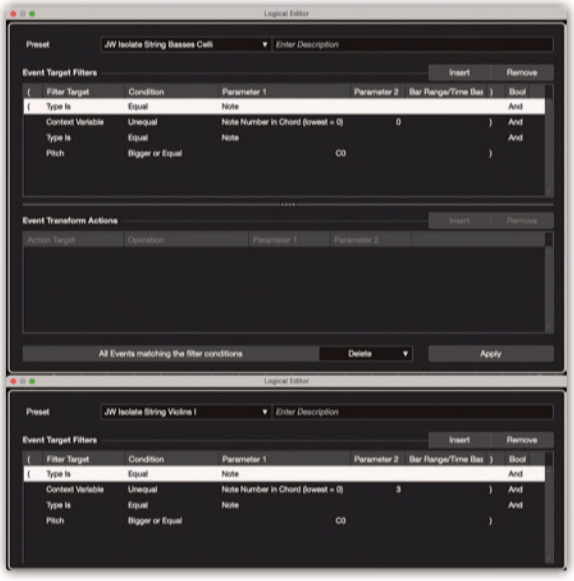
Screen 2 shows two examples of the Logical Editor presets required for this. The main part of the screen shows a preset I've named Isolate String Basses Celli. There are two filters specified in the Event Target Filters panel, and both act to select/deselect specific notes. Thanks to the Context Variable entry in the second line, the first two lines only select notes within the clip when Note Number in Chord is not zero. Zero represents the lowest note in a chord, so this selects every note within a MIDI chord except the lowest one. Lines 3 and 4 apply a second note-based filter that selects notes only if their pitch is greater than or equal to C0.

This criterion means that any keyswitches used during a performance (all of which lie below C0) will not be selected. There are no entries in the Event Transform Actions panel, but note that it is set to Delete. When these criteria are combined, then, and the Logical Editor preset is executed on one of our MIDI clips (either the Basses clip or the Celli clip) Cubase selects and deletes all notes above C0 that are not the lowest note in the chord and are above C0. In other words, we end up with a clip containing just the lowest note in the chord plus any keyswitches, meaning our bass (or cello) patch will now just play a single line from the full chord.

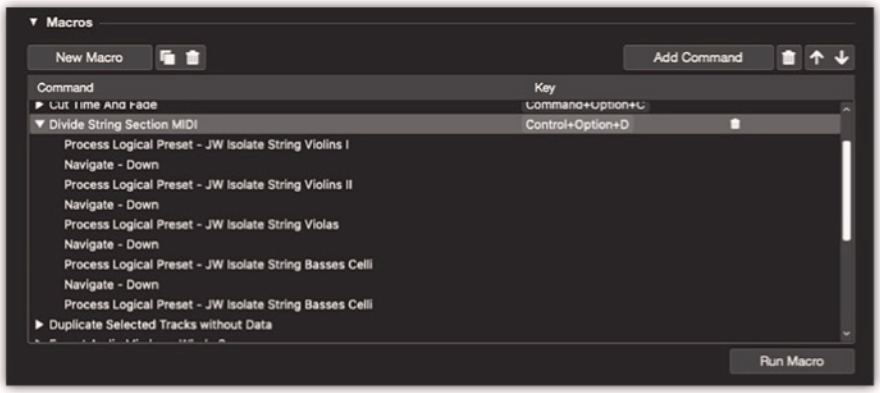
You'll need five similar Logical Editor presets in total, one for each sub-section/part. They all follow a similar pattern, but obviously some tweaks are required. As an example, the second part of Screen 2 shows just the Event Target Filters for my Isolate Strings Violins I preset (the Event Transform Actions panel remains the same). Note that the only difference is that, in this case, the filtering leaves in the fourth highest note within the chord (Parameter 2 is set to 3) as well as the keyswitches.

Make Mine A Macro

Once the five Logical Editor presets have been created, thinning out each MIDI clip simply involves selecting the clip and executing the appropriate preset. If you wish, this can be achieved from a menu: MIDI/Logical Editor/Apply Preset. This pops open a panel with all the Logical Editor presets listed, and it's very easy to step through the five MIDI clips to do what's necessary. You can also assign any Logical Editor preset to a key command. But for a faster approach, check out the Macro shown in Screen 3, which can be executed with a single keystroke. Providing you have the MIDI tracks laid out in the way suggested here, selecting the topmost MIDI clip (for the Violins I MIDI track) and executing the Macro will execute the required Logical Editor preset for Violins I, then select the next MIDI clip (thanks to the Navigate – Down command) and repeat the process. In all, it takes less than a second to transform your chord into five MIDI clips, each with the desired single-note line, and with your keyswitches remaining intact.



Screen 2: Two examples of Logical Editor presets that allow each string sub-section to play a single-note line from a four-note chord.



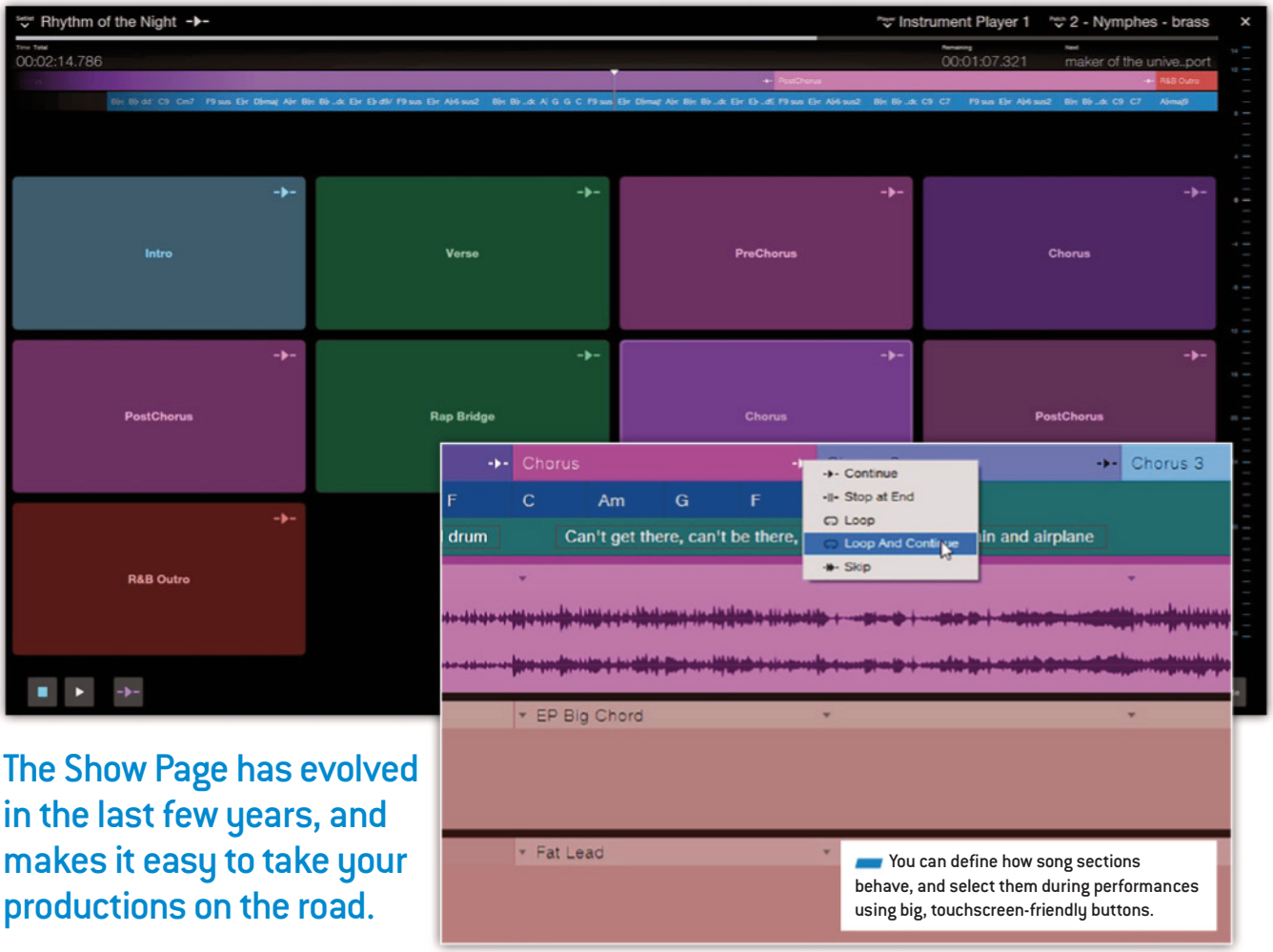
Screen 3: If you want to work fast, try chaining the various Logical Editor presets into a Macro that you can execute with a single keystroke.

Make It Real (Time)

To close, let's consider one question that the above approach raises: why can't this 'chord separation' task be carried out in real time, as you play the MIDI chords? Well, unfortunately, Cubase's MIDI Input Transformer (the real-time version of the Logical Editor) doesn't support the Context Variable option, I suspect because the unpredictable timing of incoming notes makes this sort of processing trickier to perform reliably. I do wonder, though, whether Steinberg might be able to add such capability to the (already excellent) Chord Pads system. Those trigger the whole chord at once, so in theory at least, it would be possible — in fact, it would be a very cool solution to this problem, and could also have applications beyond just orchestral strings. Pretty please, Steinberg?

In the meantime, in case you do a lot of orchestral composing, would *like* to do this in real time and have some money to throw at the problem, it's worth drawing your attention to a Mac/Windows app called Divisimate (https://divisimate.com), which you might want to check out. This software sits between your MIDI keyboard and Cubase and, amongst other things, allows you to split out the notes of an incoming live chord onto different MIDI channels.

Of course, whether a part is sustained or played rhythmically, there can be (much!) more to creating a string performance than relying exclusively on block chords. Indeed, for a convincing result, further refinement of the parts may well be required. You might try lowering the basses by an octave, or revoicing the other notes within the chord to create different inversions. Equally, you might edit the Violins I and II lines to let them 'cross' in pitch, and create more harmonic interest. Or perhaps you could let the basses, celli, violas and second violins carry the chord and add a melodic top line in the first violins. While it would be great to see a real-time option for this sort of thing in the Cubase feature set in the future, the Logical Editor approach I've described here already gives you a great starting point for creating more detailed string (or brass/woodwind) arrangements in double quick time. And if you'd like to hear what all this can sound like, check out the two audio examples we've put on the *SOS* website (https://sosm.ag/cubase-0424).



You can define how song sections behave, and select them during performances using big, touchscreen-friendly buttons.

The Show Page has evolved in the last few years, and makes it easy to take your productions on the road.

ROBIN VINCENT

Studio One's Show Page is, in essence, a way of incorporating a gig or live show manager into your DAW. The general idea is that you're a band or performer who wants to play along to a backing track, with perhaps a live vocal and guitar, and some keys playing a virtual instrument or maybe a real synth (though you can subvert that entirely, as I'll explore towards the end of this article). You can export your finished Studio One tracks directly to it, and on your tour it can handle backing tracks, live instruments and virtual instruments, let you cue up playback songs and sections, and send chords and lyrics to all your bandmates.

For live performers, then, it's a pretty useful extension to Studio One — which is why I wrote about it in *SOS* three years ago (https://sosm.ag/studio-one-0221). But a few updates have since smoothed out the experience, as well as delivering new features including the Arranger track and lyric display. So I thought now would be a good time to revisit the Show Page. I'll try to show why it's probably more useful than you think...

Show Me Around

The Show Page Overview has a familiar feel, looking suspiciously like the arrange page in 'regular' Studio One: you have the same browser, the mixer's here, and there's a timeline, as well as something that will probably hold a list of tracks. But you have to think about this space a bit differently because, rather than writing and recording music, the Show Page is a place for *performing* it — it's where you build a live performance set from completed songs or backing tracks. So, a song brought in from Studio One appears as a single, exported, mixed and processed audio file; there's no editing or sequencing available, just playback, accompaniment, and some rather nice equipment automation. If you're still writing your song, the Show Page is not for you...

In the main window (where, in a conventional DAW, you'd expect to have tracks) are what PreSonus call Players, and there are three types of Player: Backing Track Player, Real Instrument Player and Virtual Instrument Player. You build up your gig by adding the backing tracks for each song, one after the other, along the timeline. Then, you can tell it to switch presets, switch sounds, change plug-ins, and so on for each song as it comes. You can add chords for your guitarist and lyrics for the singer and control it all from the computer or an app running on a tablet. Once you're ready to play, you hit the Perform button, and it all transforms into this sleek, minimalist and large-scale control screen that contains everything you need to put your hands on your instrument and perform. Let's look at the details.

Players

Taking the general band idea as our guide, we can drag in our backing track from the browser and, underneath

that, drag in our virtual instrument and create a Real Instrument Player for our guitar. Drag and drop works as it does in conventional Studio One, so it's easy to drag in instruments, presets and audio files, and have things appear automatically.

The Backing Track Player is a simple audio track. If you've exported from a song into the Show Page, this is where your audio mix sits. You have Mute and Solo buttons, output routing at the bottom and, in the middle, everything for the Patch system that we'll get to in a minute. By the way, the Read button is also related to the Patch system rather than the sort of automation you'd normally expect in Studio One.

You can create a Virtual Instrument Player by dragging the instrument from the browser. The setup is very similar to the Backing Track Payer, but this time you have MIDI In and Out options and the Patch field is populated with the selected instrument.

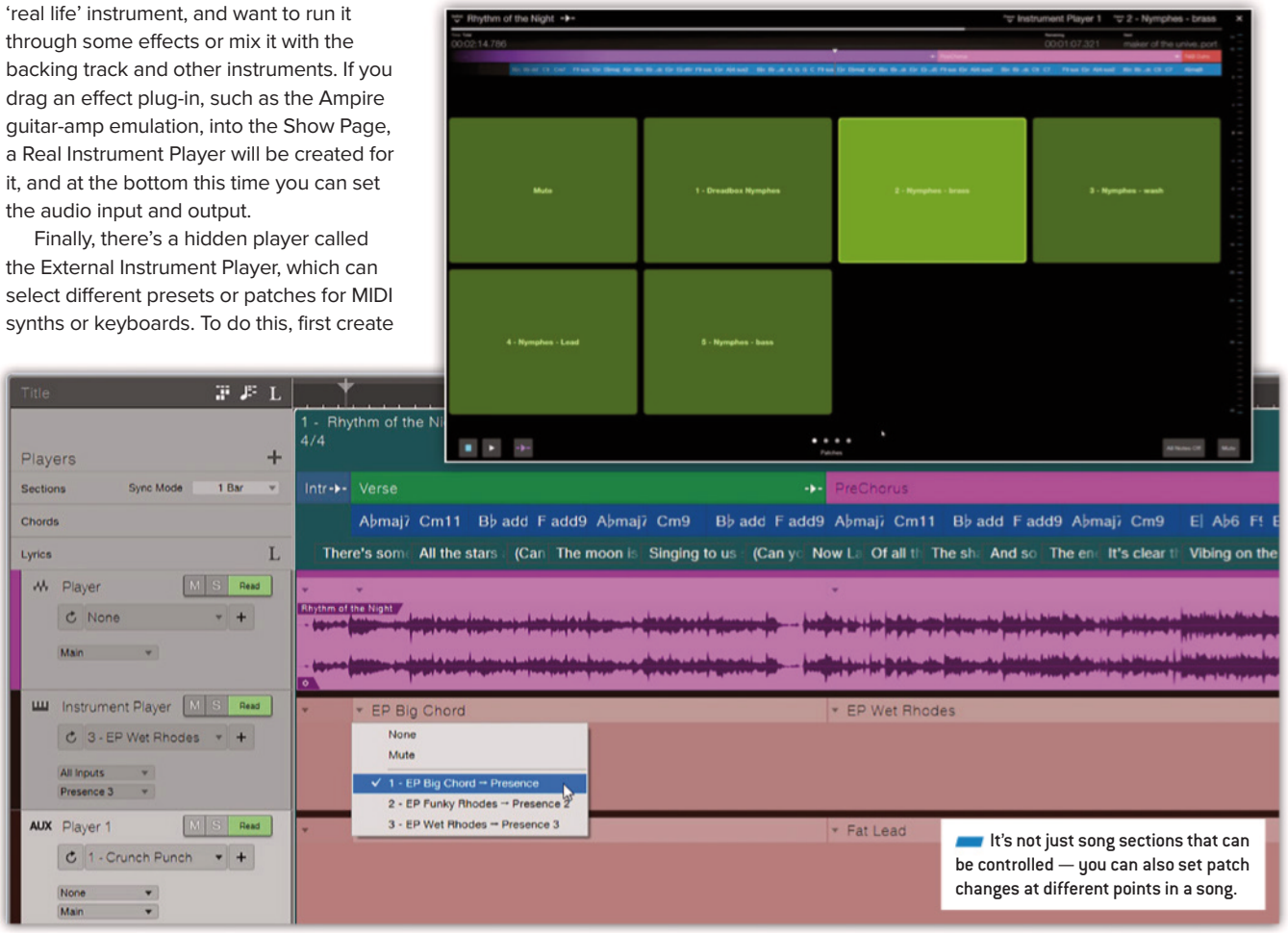
A Real Instrument Player is set up like an auxiliary input track — it even gets labelled 'AUX'. The assumption is that you're using a guitar, microphone or other 'real life' instrument, and want to run it through some effects or mix it with the backing track and other instruments. If you drag an effect plug-in, such as the Ampire guitar-amp emulation, into the Show Page, a Real Instrument Player will be created for it, and at the bottom this time you can set the audio input and output.

Finally, there's a hidden player called the External Instrument Player, which can select different presets or patches for MIDI synths or keyboards. To do this, first create a Real Instrument Player for the audio to mix it through the same system or add effects. Then create an External Instrument for your synth, using Studio One's Add Device system. You create a new instrument, give it a name, allocate it a MIDI output and so on. Once it's in External Instruments in the browser, you can drag it in to create its own Player. (It's different from a Virtual Instrument Player because it has a field for program and bank numbers.) You can also use this with MIDI-enabled effects. While these are not instruments as such, if you set effect boxes up an External Instruments, you can drag them into a Player and have the Show Page select the right effect for each song — very neat.

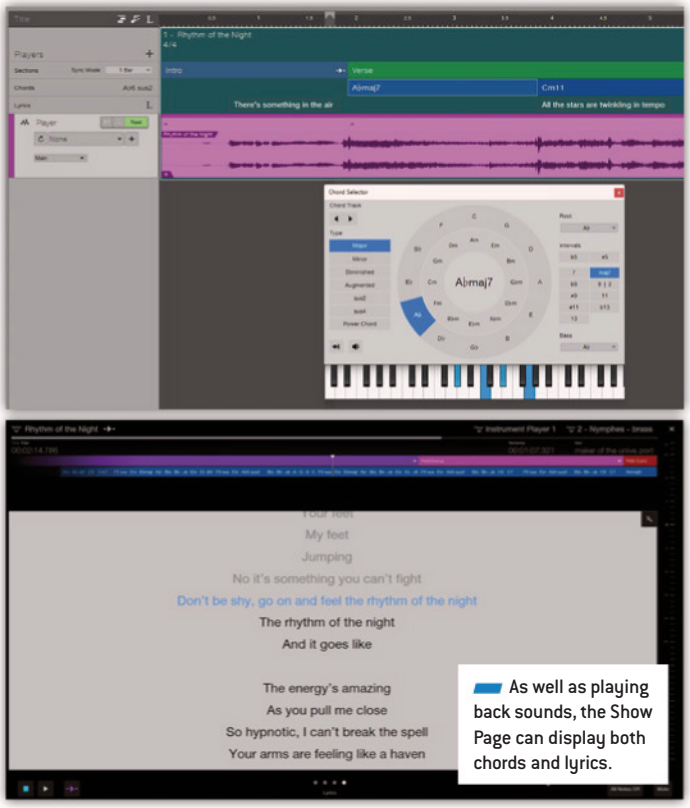
Songs & The Arranger Track

When first released, the Show Page was mostly about songs, and you can still work that way. You add your songs, and it creates a setlist. Each song can load different presets and patches for effects, sounds and instruments. Your backing track tends to form the structure of each song, and as you add them to the timeline, they appear in the setlist on the left. The header above the song gives the name, time signature and tempo. A little icon to the top right offers a couple of options. It can play the song and stop, letting you wait for applause before manually kicking off the next song. It can loop it indefinitely, if for instance you have a loop that you want to riff over until you're ready to launch into your hits. Or it can continue straight to the next song.

Since version 5 of Studio One, though, the Show Page has also had an Arranger Track. This lets you break a song up into sections like Intro, Chorus, Verse and so on. Each section has a little icon of its own and a number of additional options. Like whole songs, they can stop, loop, or continue. But they can also loop a set number of times and continue, or be skipped altogether. You can leap between sections with the double click of a mouse. How quickly and smoothly that happens is down to the Sync Mode setting, on the left. This can be set at one to four bars, or none, or you can have it leap once the current section has finished.



It's not just song sections that can be controlled — you can also set patch changes at different points in a song.



As well as playing back sounds, the Show Page can display both chords and lyrics.

» The Arranger Track is potentially transformational to the management of your show: now, rather than looping a warm-up song indefinitely, you can loop just the intro, or you can hang onto that middle chorus as long as you like while the guitarist enjoys an extended solo. In some ways, it can even become a simple clip launcher. Its versatility is awesome.

Chords & Lyrics

Once the Arranger Track appeared on the Show Page, it wasn't long before the Chord Track and the Lyric Track followed suit. You can write in lyrics for each section and add chords using the Chord wheel and Lyric Display borrowed from standard Studio One. Or, perhaps more efficiently, the whole Arranger, Chord and Lyrics tracks can all be imported from your Studio One song when exporting it to the show. Note that if you manually enter the lyrics into the Lyric Track, they'll appear in Perform mode as a continuous line of text — to avoid this, right-click on the lyric and tick the New Line or New Paragraph box, so that the Perform mode knows how to present it.

In Perform Mode, the lyrics get a whole screen to themselves for a full-on karaoke experience. The Chords, though, get relegated to a thin line at the top, where they sometimes obscure each other — hopefully something that can be improved in a future update.

Patches

The key to the fluidity in the Show Page is the Patch system. You can take a snapshot of each Player at any time: this stores the Player's settings, effects, plug-ins, loaded instrument, preset and external MIDI settings. These are saved as Patches, which can be recalled manually or, even better, automatically at the start of any song or section. So, not only will the Show Page select the right sounds for the song, but it can also switch sounds for the chorus or turn on effects for the solo. It's a beautiful thing.

To create a Patch, first set up your Player with everything you want to happen. This can be the virtual instrument with the right preset loaded, effects dropped in the mixer, mixer volume and pan, or bank and program numbers on External Instruments. Then hit the '+' button on the player header and call the Patch something helpful. Job done. Now you can make adjustments for the next section or song and create another patch. A quick way to do this with PreSonus virtual instruments and plug-ins is to drag a preset from the browser onto the Player: this automatically generates a Patch.

To set up the automation, choose the Patch from the little drop-down arrows beneath the sections or songs on the Player's track. You can also automate mutes so you can drop players out when they are not supposed to be playing, which is a great way of dealing with over-noodling guitarists!

Perform Mode

With your set list done, your Patches lined up for auto-switching effects, presets and fiddling with your external synth sounds, you're ready to perform. Perform Mode throws away the clutter and gives you a focused, action-packed, extra-large overview. The timeline runs across the top, with the chords following along. From menus at the top, you can select the song from the setlist, set which Player you want to use and choose the corresponding Patch.

You then have four views. In Patch view, huge buttons can select any of the patches for your chosen Player. In Sections view, you can switch between the different Arranger Track song sections and swap them in and out of loop mode. Lyrics view gives you a highlighted line-by-line view of the lyrics in the Lyric Track. Finally, a control section creates a virtual control surface for your devices — rather than fiddle around with the tiny controls on a virtual instrument, you can map some nice, large faders and knobs and control those on screen.

All of this comes together very beautifully in a touchscreen interface. You can use an iPad, Windows or Android tablet along with PreSonus' Studio One Remote software to mirror the Show Page Perform view and access everything with your fingers. It's perfect for a live performance environment, where you definitely don't want to be scrabbling around for a mouse!

Other Uses

Everything about the Show Page feels designed for gigging bands, worship setups, weddings and entertainers. But it also has the potential of reaching some other creative opportunities. You could, for example, use it as a karaoke machine, with backing tracks and accompanying lyrics on a nice big display. Or you could ditch the backing tracks altogether and simply use it as a studio gear manager. And once you start dropping audio loops into the Players, it has potential as a rudimentary clip launcher, along the lines of Ableton Live or Bitwig Studio. There's no automatic time-stretching as yet, so you'd have to pre-prepare your loops, but looping sections and cueing up new ones is super easy. The other big one could be in theatre, radio or podcasting, where you might want a scattering of sound effects, atmospheric music, background sounds, and so on, which can be cued up and launched precisely when you need them, or looped as long as you want. I sense there's a lot more to come with the Show Page...

soundtoys

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Spotlight

Three-way Monitors

Three-way studio monitors usually come at a premium. We round up the models that won't break the bank!

LUKE WOOD

The vast majority of nearfield monitors are two-way designs, in which the midrange reproduction is shared between low- and high-frequency drivers that become less efficient as they move toward the limits of their bandwidth capabilities. Adding a dedicated midrange driver combats this, allowing each of the three drivers to focus on a narrower frequency range. And although a three-way speaker requires an additional crossover, the two crossover points and their associated phase distortions can be moved away from the all-important midrange. But these benefits come at a cost, so in the past, three-way monitors have mostly been found towards the top end of the market. Though they are still more expensive than two-way speakers, in recent years their price has been coming down, and this month we're shining the *SOS* Spotlight on a selection of three-way speakers that are available for under $5000 per pair. All prices given below are per pair.

ADAM Audio A77H

ADAM A77H & A8H

Announced in 2022, ADAM Audio's A Series replaced their popular AX Series and introduced some significant new features. A Multi-Layer Mineral (MLM) low-frequency driver design, new magnet system and improved cabinet design enable them to offer a high output level with minimal distortion, and onboard DSP implements a wealth of room optimisation features, including support for Sonarworks SoundID Reference calibration files. All of the A Series' DSP functionality can also be controlled remotely using ADAM's A Control Mac and PC software. The range includes two three-way designs. The smaller of the two, the A77H, features a pair of seven-inch MLM low-frequency drivers, along with the Dome Cone Hybrid (DCH) midrange driver found in the S Series units. The A8H, the largest member of the A Series, employs eight-inch MLM and 3.5-inch DCH drivers for low- and mid-frequency duties. In both models, the high end is taken care of by an X-ART 'ribbon' tweeter, and the company's HPS (High-frequency Propagation System) waveguide ensures wide, consistent horizontal dispersion while keeping the vertical dispersion narrower. Helpfully, this tweeter can be rotated, so the speaker can be positioned horizontally or vertically. Rear-panel controls provide access to a four-band EQ that offers over 100 tuning options in preset steps, as well as a Voicing selector that offers a choice of tonal balances.

$ A77H $2599.98. A8H $2999.98.
W soundonsound.com/reviews/adam-audio-a8h
W adam-audio.com/en/a-series

Barefoot Sound Footprint03 & Footprint02

The latest additions to Barefoot's range aim to deliver the renowned sound quality of their Footprint speakers in a more compact enclosure at a more affordable price. The Footprint03 introduces a new bass reflex design with a side-firing port. Coupled with high-quality cabinet construction, this allows the monitor to deliver a low-end extension (-3dB at 45Hz) that the company say is comparable to that of their larger sealed-cabinet designs. The low- and mid-frequency drivers pair a rigid aluminium cone with a high-sensitivity motor, and feature additional copper shorting rings around the voice coil, to help minimise harmonic distortion. A newly designed one-inch ring radiator tweeter offers longer excursion capabilities, with increased output levels and reduced compression. Onboard DSP enhances the low-end response and distortion figures further, and Barefoot have implemented a new SPOC (SPectrally Optimised Conversion) technology that uses an analogue crossover section before the A-D stage — this allows high frequencies to be passed on unattenuated, resulting in an improved signal-to-noise ratio. These speakers also feature the company's MEME (Multi Emphasis Monitor Emulation) technology, with four switchable voicing options that can be selected using a remote control. The larger Footprint02 is the next step up in the range, a sealed design that utilises the company's Dual-Force subwoofer technology, with a 6.5-inch low-frequency driver on each side of the cabinet.

$ Footprint03 $1995. Footprint02 $2995.
W barefootsound.com/footprint03
W barefootsound.com/footprint02

Dynaudio LYD 48

Dynaudio's LYD range aims to provide a serious mixing setup in a relatively compact and affordable footprint, and was developed using the knowledge gained from supplying large reference systems to professional studios. But they also aimed to create speakers that look as good as they sound, drawing inspiration from their range of hi-fi systems. The three-way LYD 48 is the largest model in the range. A rear-ported bass reflex cabinet houses Dynaudio's proprietary MSP mid- and low-frequency drivers, along with a newly designed soft-dome tweeter. The -3dB points in the frequency response sit at 50Hz and 21kHz, and the monitors promise to deliver a neutral and accurate sound that remains consistent whatever the level they're putting out. A built-in DSP controller makes it possible to extend or curtail the low-frequency response, apply bright, neutral or dark tilt EQ settings (with linear or minimal phase filter options), and choose between Wall or Free modes, to tailor the output of each monitor according to its placement in relation to boundaries.

Dynaudio LYD 48

$ $2398.
W soundonsound.com/reviews/dynaudio-lyd-48
W dynaudio.com/professional-audio/lyd/lyd-48

EVE Audio SC3070

EVE Audio's SC3070 was designed to deliver the midrange reproduction of the SC4070, a model renowned for its balanced sound and low distortion figures, but in a much more compact cabinet that can be used in smaller rooms. As with all of the company's monitors, high-frequency reproduction is taken care of by an AMT (Air Motion Transformer) driver; here, this is joined by four-inch and seven-inch fibreglass-coated SilverCone drivers that promise to deliver an accurate midrange and a precise and powerful low end. The analogue inputs feed Burr-Brown A-D converters, with the SC3070's built-in DSP catering for everything from crossover duties to overload protection, including adjustable low-shelf, midrange and high-shelf filters which the user can use to tailor the output to their room. A front-panel SMART knob provides easy access to volume and filter settings, but selected settings can also be fixed using rear-panel DIP-switches to prevent unintentional (or unwanted) adjustments.

EVE Audio SC3070

$ $3798.
W eve-audio.com/index.php?page=SC3070

Fluid Audio Image 2

Fluid Audio say that they've pushed the limits of speaker design with the Image 2, which delivers accurate stereo imaging, a flat frequency response, impressive low-end extension and low distortion figures. The design employs a pair of side-firing eight-inch low-frequency drivers that are interlocked at their magnets to help improve cabinet stability and low-end response — this is just -2.2dB down at 40Hz and features a natural-sounding 6dB/octave roll-off, »

Fluid Audio Image 2

» thanks to the sealed cabinet design. A five-inch aluminium-cone midrange driver boasts a motor structure that has been optimised to reduce harmonic distortion, and a newly designed AMT driver handles the high frequencies, promising a precise, detailed response with plenty of headroom and low distortion. The monitors' performance is aided by onboard DSP, which is also capable of hosting up to four Sonarworks SoundID Reference room-correction profiles — these can be transferred and selected using Fluid Audio's DSP Configuration Tool software. Along with the analogue inputs, the Image 2 also features AES3 and S/PDIF connectivity, allowing those with spare digital outputs to connect to the monitors with fewer stages of conversion.

$ $3798.
W soundonsound.com/reviews/fluid-audio-image-2
W fluidaudio.com/product/image2

HEDD Audio Type 20 Mk2

Described by HEDD as a perfect all-rounder, their Type 20 Mk2 combines impressive technical performance with some innovative features, including the ability shared with other Type 2 Mk2 models to operate either as a sealed or ported cabinet — each monitor comes supplied with a plug that can be fitted to the port, and a rear-panel switch makes it possible to select between closed and ported modes. The Type 20 Mk2 employs the company's renowned high-quality AMT drivers, which have been designed to deliver fast transients along with an extended high-frequency response, to ensure it delivers a natural-sounding top end with minimal distortion. As well as providing shelf filters and low-frequency range adjustment, the monitors' onboard DSP includes a Lineariser function, a system HEDD have developed to improve transient response and stereo imaging by tackling the small time delays inherent in loudspeakers. The Type 20 MK2 can accept analogue line-level or AES digital inputs.

$ $4798.
W hedd.audio/products/type-20-mk2

Kali Audio IN-5 & IN-8

Kali Audio's IN-series features two monitor models, the IN-5 and IN-8. Both are equipped with a coaxial driver that pairs a four-inch midrange unit with a one-inch textile-dome tweeter, helping to deliver realistic stereo imaging and promising to reveal subtle details in a mix. The IN-5's low end is then taken care of by a five-inch driver, whereas the IN-8 employs an eight-inch driver, offering a slightly more extended low-frequency response and higher maximum SPL.

Kali Audio IN-8

Kali say that their designs allow the INs to enjoy all the benefits of traditional three-way and coaxial systems, but without any of the drawbacks associated with either. Given the keen pricing, many could see the IN-series offerings as a tempting choice for immersive monitoring setups and Kali recognise this — they offer a wide range of wall and ceiling mounting accessories, including some that will connect directly with Triad-Orbit's hardware.

$ IN-5 $698. IN-8 $798.
W soundonsound.com/reviews/kali-audio-5
W soundonsound.com/reviews/kali-audio-in8
W kaliaudio.com/independence

Neumann KH310A

Since speaker manufacturer Klein + Hummel were acquired by Neumann, a number of their popular designs have been updated and released as part of the KH series. One such example is the popular O300, which lives on in the form of the KH310A, an all-analogue three-way monitor designed to provide high-end performance in a compact enclosure.

All three drivers are designed in-house by Neumann using their own software. The low-frequency unit is constructed from a composite material, and is joined by fabric-dome midrange and high-frequency units. The internal cabinet construction is optimised to combat standing waves, and the front panel designed to reduce diffraction, while an MMD (Mathematically Modelled Dispersion) waveguide surrounds the tweeter, for a smoother off-axis response, wide horizontal dispersion and narrow vertical dispersion. A trio of rear-panel Acoustical Control switches provide some EQ to tailor the response to suit different rooms, and it's possible to pair these monitors with Neumann's KH750 subwoofer — as well as extending the system's low-end response, this adds some DSP-based room correction.

$ $4990.
W soundonsound.com/reviews/neumann-kh310a
W neumann.com/en-en/products/monitors/kh-310-a

Neumann KH310A

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Photo: Lee Kirby

Hans Zimmer & Friends

For the *Dune* movies, Hans Zimmer and his team of extraordinary musicians didn't just create a score. They created an entire world.

SAM INGLIS

"When I was a kid and I went to see a science fiction movie," remembers Hans Zimmer, "it would be set in a galaxy far, far from here, thousands of years in the future — and then I would hear the string section start. I'd go, 'Really? Is that it? Nothing new has happened since then?' Why would I have a romantic Northern European orchestra? Let's go and invent instruments. Let's go and build things!"

The composer has done just that on his scores for the *Dune* movies. Based on Frank Herbert's classic sci-fi novels and directed by Denis Villeneuve, the second instalment is about to hit

Tina Guo and Guthrie Govan take a break during sessions at Remote Control.

Woodwind maestro Pedro Eustache holds the chromatic duduk he created in order to play one of the main themes from *Dune*.

cinemas worldwide at the time of writing, with musical ambition and creativity dialled up to 11.

Other Worlds

"It starts off," says Zimmer, "with creating a sound world. One of the things that I've always done is that I spend a long time looking at the colour palette and talking to the director of photography. What's this going to look like? What colours are you going to be using? What's the colour palette of this planet? And that seeps over into the colour palette of the music."

Creating a sound palette for *Dune* meant assembling an extraordinary team. Vocalist Loire Cotler, guitarist Guthrie Govan, flautist Pedro Eustache, bassist Juan 'Snow Owl' Herreros and cellist Tina Guo are all long-term collaborators and soloists in his World Of Hans Zimmer and Hans Zimmer Live spectaculars. Their sonic contributions to the first *Dune* movie were mostly remote, as production took place during the Covid pandemic. Much sampling took place, often with the idea of creating otherworldly or alien substitutes for conventional instruments.

"For *Dune: Part One*, our string section is Guthrie," laughs Hans. "It starts off with an idea, and I go to the musicians with that. Like, 'Hey Guthrie, can you sample your E-Bow guitar so that it becomes my string section?' or 'Pedro, can you just go and make me the sound — in pitch, in tune — of the wind whistling through the desert?' So there's a lot of sampling going on, and then there's an enormous amount of creating sounds going on."

"Hans wanted his 'string section' orchestrations to sound somewhat otherworldly," expands Guthrie Govan, "so I tracked a lot of those string parts using slide guitar, fretless guitar and E-Bow.

"We were touring in Europe when Hans requested raw materials for an E-Bow-based sample instrument, so it became part of my hotel room ritual to set up a Fractal FM-9 processor and record as many long notes as I could bear. For the sake of my sanity, I had to divide the recording process into multiple sessions! I think I must have recorded something in the region of eight passes for every alternate note on the instrument's register, doing my best to find a balance between a touch of vibrato and the required stability of pitch — a strangely hypnotic process, as every note needed to be eight bars long! I was pleasantly surprised by the sound of the finished instrument.

"We also made more extensive use of feedback on this new score [*for Dune: Part Two*]. Whilst on the road, we would set up in one of the backstage areas and run my Fractal FM-9 through a powered monitor cranked up to obscene volumes! I discovered that something fun happens if you record guitar feedback with a high-gain overdriven tone but also capture the DI guitar signal and then reamp it through a clean amp setting... the result is a haunting noise which could never have occurred in real time."

On The Edge

"I think what we're going for, ultimately," says Hans, "is the extreme of the instrument. What is it that doesn't exist out there in the world? The one thing we have in common is, you know, how can we make a sonic world that hadn't existed before? So by the end of it, we create an orchestral palette which is not the palette of the normal Western European orchestra."

Key to making this work is that the musicians Hans works with are not only virtuosos in the conventional sense. They're also sonic experimenters, who are keen to explore the outer limits of what's possible with their instruments. "Geniuses like Hans don't care about what's possible or not," says Pedro Eustache. "They have a vision, they have a mission, and it's up to us to accomplish that."

On *Dune: Part Two*, fulfilling Hans' vision certainly required the impossible of Eustache. For the Hans Zimmer Live tour that followed Part 1, the composer »

Photo: Semmel Concerts / RCI Global / Frank Embacher

Loire Cotler's extraordinary vocals have become a core part of Hans Zimmer's live performances, as well as featuring heavily on scores like *Dune*.

Electric instruments recorded acoustically played a major role in the *Dune* sound world.

» decided to start the show with Loire Cotler alone on stage, singing one of the main themes. "She starts singing very quietly as she walks onto the stage and goes through the whole length of the stage," explains Pedro. "And Hans is crazy enough that for *Dune: Part Two* he goes 'Pedro. I want you to play the theme on the duduk.' I'm like 'Are you crazy? That's not in the duduk's range.' So what am I gonna do?"

What Eustache did was to invent a new instrument: the chromatic duduk. "Thanks to Hans challenging me, I had to come up with stuff that does not exist. Which enriches me and makes me even more unique because then I have access to things other people don't have."

"The material I've been involved with for most of my musical career is considered at the extreme of vocal repertoire," says Loire Cotler. "So I view these seemingly impossible vocal tasks that Hans is asking as an opportunity to take me farther in a direction that I've been passionate about for many years. When Hans' musical ideas are on fire and he starts gives us some outrageous directions... the only course of action is to grab our instruments by the guts, and leave our bodies behind!

Hans Zimmer: "It's not every movie where you get to unleash the synthesizers and the electronics in such a way!"

"I did quite a bit of intimate, close-miked, gentler, inside-your-ear whispery rhythms and breathy rhythmic hits, where I emulated a shakuhahchi hit, inspired by Pedro of course! Another impossible Hans assignment was to create some 'white noise grooves' using vocal percussion, and then track them up in a bunch of meters. And then of course the more traditional singing, like in the opening and closing of 'House Atreides' on *Dune Sketchbook*, where I'm practically kissing the mic."

"Hans is always very enthusiastic about the idea of pushing an instrument's boundaries," agrees Guthrie Govan. "I remember seeing a little twinkle in his eye when I brought an eight-string guitar to the rehearsals for our first tour together. The next time I saw him in his studio, he had treated himself to a nine-string!"

Up Close

"For me, there was no organic cello," says Tina Guo, "because we're on an alien planet and they didn't have cellos. So we thought, OK, maybe we'll use the electric cello — and in addition to getting the signal from the instrument, we miked the actual electric cello. It sounds terrible, but it's terrible in a good way, in a very weird way where it's like 'What is that!?' There's a lot of weird little sounds, because it's not supposed to be played that way, but it actually sounds really interesting. It's barely audible, so if anyone even breathed, we're like, 'Oh God, we have to start over again,' because you could hear everything in the room."

"There was a lot of close miking on the body of instruments that would normally be amplified," agrees Snow Owl. "They can sound pretty dead, but with the right microphone combination and the right placement we kind of discovered a secret world, different parts in the body of the instruments that resonate in a very unique way.

"Another thing that I did which Hans lost his mind about was to put magnets all over the body of the bass so that it turned the strings into oscillators. It became self-oscillating and then I was just honing on different parts of the body. According to Hans, this captured the complete isolation of the »

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Originally from Colombia, bassist Juan 'Snow Owl' Herreros now lives in Vienna, where he has his own recording studio.

» desert vibe, and he really used a lot of that in the score.

"During the creation of the first *Dune* there was a certain bass sound that I created that became really associated with the Harkonnens. Hans really wanted to exploit that on this second part, so we did hours and hours of sampling the sound so that Hans and the rest of the team could compose with it. And then when they put all the stuff together, then I came to LA and I recorded all the ideas that they had, kind of like building the bridge between the samples and the real person performing it. It was hundreds of hours of sampling, creating the craziest sounds on multiple basses, and using different microphone techniques and everything to sound like we're not on Earth any more.

"It was always about: how can we tell a story and how can we attach it to certain characters? These sounds ended up becoming sound design for characters and for parts of the world, even some of the spaceships. It's not just meant to be the low-end fundamental. it's a melodic element and a thematic element. I used specific Mongolian bows on very hard strings and it sounds very rough. It was very difficult to bow notes out of it. But it fits so well to the darkness of the black and white monochrome world of the Harkonnens."

Guthrie Govan's approach was equally unconventional: "My entire toolkit for the *Dune* movies was essentially a Fractal FM-9 processor, my signature Charvel guitar and a Vigier fretless electric, coupled with a slide, an E-Bow and a volume pedal. The Fractal gear is perhaps best known for its amp-modelling algorithms, but its effects processing capabilities are also hugely flexible — pretty much anything can be used to modulate anything else, and the signal chain lets you run up to six chains in parallel, so its uses go way beyond the obvious 'virtual pedalboard' applications. One of the more fun things I concocted for this score was a patch using microtonal clusters, with the pitches spread out in increments of 0.75 semitones. A patch like that forces you to think more in terms of pure texture and sound design — it's basically impossible to sound melodic

The Synths Of *Dune*

"It's not every movie where you get to unleash the synthesizers and the electronics in such a way!" laughs Hans Zimmer. Both Remote Control in Santa Monica and his home from home in London are chock full of enormous modular synths. In practice, though, time usually dictates that even on a largely electronic score like *Dune*, they play a secondary role to software instruments.

"I'd love to have all the time in the world and go back to modular, but the big difference about doing a film score now, as opposed to doing a film score 20 years ago, is the edit. There is never a locked picture. The picture's always in flux. So you need to always be able to go back to the sound, which is a little difficult when somebody tidied up your patch cords!

"The main synth I use, the synth I have never gotten bored with, is Urs Heckmann's Zebra HZ, the dark Zebra, and then I use a thing called Legend, which is basically a recreation — no, it's more than a recreation — of a Minimoog.

"And the virtual world is really good. Things sound really good. You know how important, for instance, film is to Christopher Nolan? He's never said to me that I have to go and record on tape. The last movie I recorded on tape was *Gladiator*. And I lost years of my life while the tape was winding back.

Hans Zimmer's London studio is crammed with analogue synths, including this Roland System 700 and EMS VCS3 — Zimmer's first ever synth, which he sold many years ago and recently tracked down and bought back.

"We need to embrace the technology, and it's great. I think the advances in film music are partly the advances in technology. And it's not like the technology is driving us. I think it's more that the stuff I've been hearing in my head I can now do. The imagination is limitless, and the people who've been writing good software are great imagineers."

Ryan Rubin: Music Editor

The role of Music Editor is underappreciated, sometimes undervalued, but vital to the production of any film score. That's especially so with a project like *Dune*, where the score is created collaboratively and is constantly changing. It's a unique job that is in equal parts administrative, technical and creative.

Ryan Rubin has been Hans Zimmer's editor for more than a decade. It's his job to distil order from chaos, ensuring that the maelstrom of ideas floating around Remote Control coalesce into functional cues, and that everything gets completed on time. He's also responsible for the huge organisational challenge of managing all the files that are associated with such a mammoth project. Avoiding leaks, in particular, is a crucial issue when cues and parts are being constantly sent around the world to be worked on remotely. But it's also part of Rubin's remit to spot that a particular composition might work better in a different part of the movie, or to rebalance the stems and give it a totally different feel.

"We're doing everything kind of all at the same time," says Hans. "People are making instruments, we're writing cues. There are times where we've done a bunch of recording before we've even played a cue to the director. And then, if then we play it for them and they like it, then we go one step further. So there's this very iterative process where we keep adding and adding. We're working with people from all over the world. The only people who are gonna be there for all the time zones to have a chat with everybody are myself and Ryan, you know? So we don't get a lot of sleep."

"It's great that we're international in that regard, but it's not good organisationally speaking, because we're all dealing with people in different time zones constantly," agrees Ryan. "We're all over the place. But also we can have a meeting in Los Angeles at noon and then by the next day things will have gone out to everybody all over the world and they will come back and we can present them. I remember there was something on *Dune: Part One* that we played for Denis [*Villeneuve*] and he said, 'Where did this come together from?' And I said, 'I don't know. This literally just came in 10 minutes ago. We're all hearing this for the first time.'"

"I don't ever look at my watch and I don't ever look at my calendar," laughs Hans. "And there comes a point where Ryan gets very, very pale. And he goes, 'The dub is starting. And if we don't have music on the stage for the dubbing crew, we will all never, ever, ever work again. And our careers will be in the toilet.'"

Ryan Rubin with engineer Eva Reistad at Remote Control.

when all of the notes available to you sound so disgusting and disorientating!"

Free Expression

For *Dune: Part Two*, remote sessions were followed up with a whirlwind five-week period at Zimmer's Remote Control facility in Santa Monica — where the composer added an extra dimension. Fascinated by the possibilities of the Eagan Matrix synth engine and the instruments that use it, such as the Expressive E Osmose and Haken Continuum, he decided to go straight to the top, bringing in its inventor Edmund Eagan and sound designers Guillaume Bonneau and Christophe Duquesne. "It's not just about: you go and buy that thing and play it," explains Hans. "I always try to get the person who built it or the guy who understands it the best to come in."

"Hans' original idea was to use the Osmose with his synths," says Guillaume, "but then he discovered that with the internal sound engine you have much more interaction and much more of an 'acoustic' feeling to the sound. So »

The Eagan Matrix team at Remote Control. From left: Christophe Duquesne, Edmund Eagan, Hans Zimmer and Guillaume Bonneau.

» he asked Christophe and myself to be involved directly in the project."

Involving the Eagan Matrix experts allowed the score to exploit a forthcoming feature that is, as yet, unfinished and unreleased: resynthesis. "Basically you take a recording and make an analysis of it, meaning you're not going to replay it as a sample, but recreate the sound with a bank of sine oscillators," explains Christophe. "That's a technique that was already used in the old Synclavier and Fairlight, but that was kind of lost in between. And the technology's a bit more mature these days, so we pushed it a step higher. I personally did use a lot of analysis for the movie based on the voice of Molly Rogers, who is the violinist for Hans, and Loire Cotler who did the famous banshee cry in *Dune: Part One*."

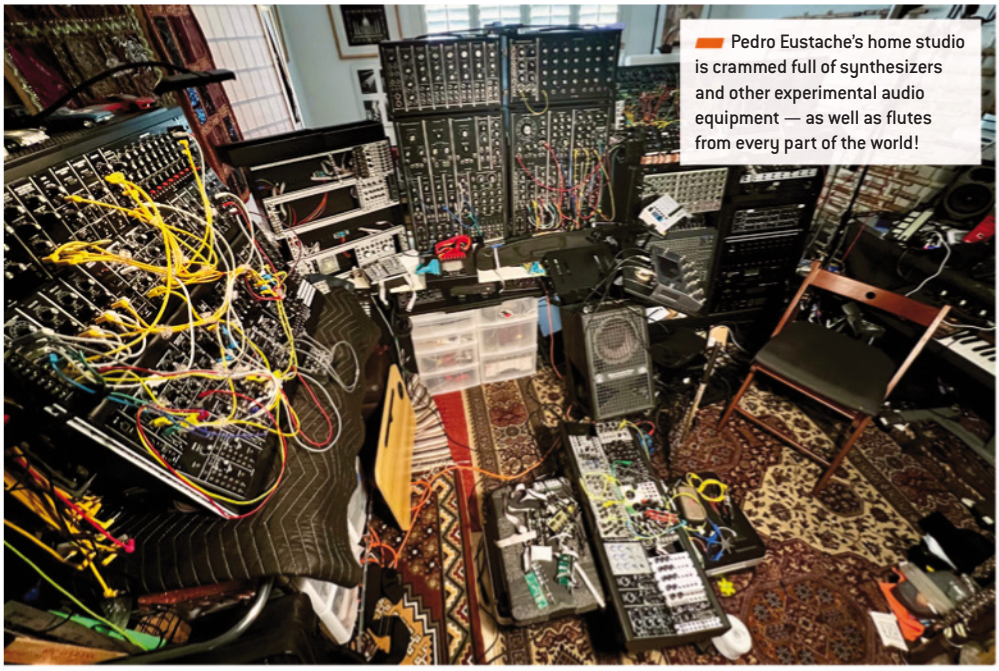
"Hans and Christophe Duquesne asked me to give them a track in character with some of the *Dune*-inspired language chants I was writing," recalls Loire. "There were specific parameters of pitch and tempo. Plus, I was in great company with the amazing Molly Rogers, who was also put to the task. When Hans and Christophe shared demos with me, I remember exactly what I said in my reply after hearing it: 'I am officially bewitched! The Voice I hear in this demo, I'll do anything she tells me to do!'"

Voices Of The Future

Guillaume: "Voices have a very strong presence in the film. So the idea is how we can change those voices that we already recorded to have another »

Everyone's An Engineer

All of the soloists who work regularly with Hans are also notable for being fascinated by the possibilities of studio recording and electronic processing. Snow Owl, for example, has a well equipped studio in Vienna, whilst Pedro Eustache's house in California is home to an extraordinary array of equipment. As well as wind instruments from every corner of the globe, he has a large collection of synths and is a devotee of wind controllers, especially the original Lyricon from the 1970s. "One of the flutes I use to this day, I bought in 1978. On the same trip, I bought my first analogue synth, a Korg MS20, which is still here in my studio. It's never been serviced and it works perfectly. Controversially, I'm a big follower of Behringer, so I've got a lot of equipment by them, but incredibly modified so my machines are not normal!"



Pedro Eustache's home studio is crammed full of synthesizers and other experimental audio equipment — as well as flutes from every part of the world!

Tina Guo's electric cello rig, meanwhile, has evolved continuously. "At the start of my career, I had every pedal you could imagine, because I thought it looked impressive, even though I only used probably four of them. And I had giant Engl amps, multiple ones. And it started getting out of hand because I couldn't even lift anything myself. My front of house guy for my last solo band project worked for Line 6, and he was one of the guys that engineered and created the Helix.

"I had tried all kinds of digital processors before and I didn't like them, but I soon realised this was because I just didn't know what I was doing. So he helped me, and showed me how to customise the sound. For the first *Dune* and the second movie, I created custom patches: Dune Fantasy, Dune Aggression, Dune Sex, like all kinds of weird things that felt like they belonged in the musical atmosphere."

"The bulk of my recording and creative work has been remote," says Loire Cotler. "I've experimented with many studio mics over the years, but I always come back to the Neumann M149, more specifically the 75th Jubilee edition. I came to know this mic when I first started working with my engineer of 18 years, Kostadin Kamcev. It has such a silky, warm, organic, midrange fullness and captures all of my extended techniques like overtone singing, just as it can translate the crisp precision of vocal percussion, and sounds really gorgeous for all the more unusual hybridised expressions."

Pedro Eustache and Edmund Eagan at work on *Dune Part 2* with a wind controller — and a lot of patch cables!

SOUND ON SOUND

PODCASTS

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Mic Polar Patterns

Understanding Which Microphone To Use

In the first of this two-part series, David Mellor gives us an introduction to mic polar patterns, explaining the best type of microphone to use for various recording situations.

Catherine Marks - Mix Engineer

The MixBus Interview

Catherine Marks talks to Kevin Paul about her traditional route into the industry as an assistant to Flood and Alan Moulder, plus her tips for how to approach a mix and get the best out of a track.

Afrodeutsche

A Journey Through Music

Afrodeutsche talks to Caro C about her musical journey, from her introduction to the music industry in Manchester, finding her sound, getting signed to Skam Records and becoming a BBC 6 Music DJ with a prime-time Friday evening slot.

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A feast of Eagan Matrices: from top, Haken Continuum Mini, Continuum Fingerboard and Expressive-E Osmose.

kind of feeling — more in the future, more spatial, something that could be something human but that we never heard. Resynthesis is a new part of the DSP that is currently in development, and so it was the best opportunity to try to use these new tools for this movie. You can keep the formant of the voice intact when you play with it, so you have a more human aspect.

"On the Osmose or the Continuum you can change the volume of the voice directly with the pressure of your finger. Your finger is the envelope — you don't need any ADSR — and you can change the shape of the sound, so you can change the volume, or create a small or huge vibrato. You can also change different parts of a single performance. For instance, you can travel to different parts of a word directly with your finger on the Continuum using a gesture, and on the Osmose by the aftertouch or any user parameter. So you can form real words directly with your gesture, but keeping the intention and human performance you get in a real voice."

It wasn't only the sound design skills of the Expressive E team that Hans Zimmer drew on for *Dune: Part Two*; they were also deeply involved as performers. "They were played on the Osmose and the Continuum, and really using the full expression of the instruments takes time," says Christophe. "The Osmose was quite new, and there was not enough time for Hans and his team to learn and practice the instrument. so he asked us to perform the sounds we were designing. We were creating the sounds, having an idea of what could work for a cue, recording it, submitting it to the team. Then the team would say 'Yeah, that's good' or not."

Although Zimmer's goal was to create an otherworldly sound palette, he nevertheless didn't want it to be overtly electronic. Many of the sound sources were thus originally acoustic, albeit heavily processed. And many of the sounds produced by synthesizers were recorded acoustically, often using physical resonators to modify sound in the live room at Remote Control.

"We had every single instrument with an Eagan Matrix inside," explains Christophe, "the Osmose, the Continuums and the Eagan Matrix Eurorack module. They were all generating sound that could be sent to the resonators, so to Onde and Pyramids from Le Voix de Luthier, also some Metallik resonators we're doing with Eowave, another French brand. So the Eagan Matrix was played through these resonators and then recorded."

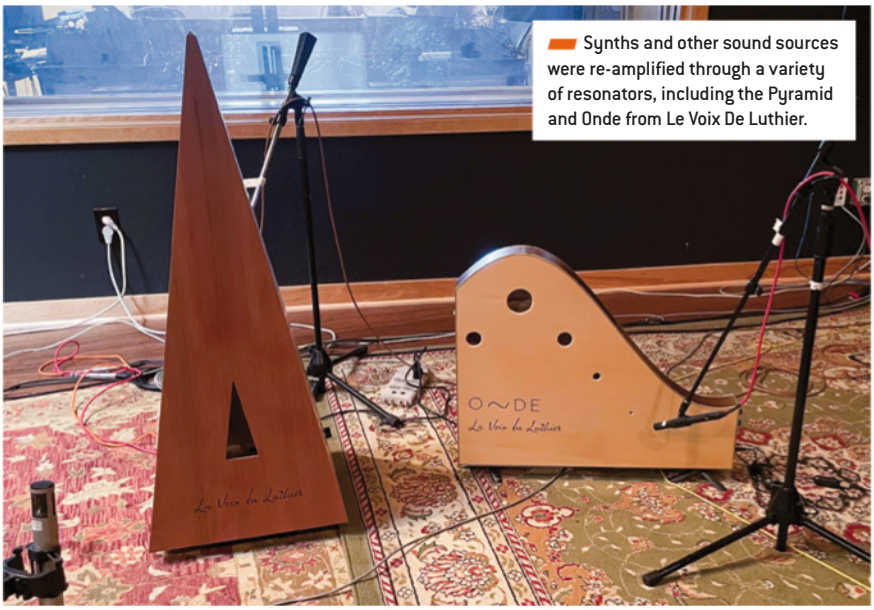
Pipe Dreams

"What you hear," says Hans Zimmer, "is not necessarily what you see! One of the fabulously cool things in the first *Dune* movie was that instead of having fanfares for the arrival of the royal family on the new planet, Denis [*Villeneuve, the director*] had bagpipes. And I thought 'That actually works, because it's timeless.' Then, at a screening in New York with a Q&A from the audience, Denis was sitting next to me, and somebody asked me about the bagpipes. I said, 'Actually they're not bagpipes. It's Guthrie Govan imitating the bagpipes on his guitar.' And Denis goes, 'My beautiful bagpipes are not bagpipes? They're guitar?' I'm going, 'Yeah, I'm sorry, I forgot to tell you. But, you know, it sounds good, doesn't it?'

"We carry on with the Q&A thing and he stops and he goes, 'Are there any other things you need to tell me?...'"

Asked how he faked the bagpipes so convincingly, Govan explains: "My solution for the bagpipe challenge was to use the monophonic synth block in the Fractal FM-9 and to layer the parts pretty extensively, varying the EQ, waveform and, subtly, the tuning for each pass... I also discovered that there was a specific way to strike the string which seemed to add an extra layer of authenticity, whereby the initial attack of the note would 'confuse' the synth's pitch tracking and add a little high-pitched warble before settling into the desired note. That all happened during the lockdown, so I had plenty of time to experiment! I remember actually fooling myself when I finally went to see the film: when that section of the film happened, I initially assumed that what I was hearing was real bagpipes!"

"The goal with those kind of new resonators or amplifiers is to keep the real acoustic behaviour of an instrument," says Guillaume. "Because it resonates through wood or metal, and then you can capture it with different kind of mics, you have a very huge stereo image. It's like being in front of someone playing the guitar, or something like that. It preserves the acoustic and organic aspect of the sound, so you don't think it's a synth, because it's the same kind of experience as listening to a cello or something like that."



Synths and other sound sources were re-amplified through a variety of resonators, including the Pyramid and Onde from Le Voix De Luthier.

Working Together

Hans Zimmer's approach to composition has always been a collaborative one. Involving top-level musicians and sound designers helps to keep his work fresh. "I'm not being elitist about it," he says, "but it's really important to me that I don't want to hear our sound in somebody else's movie."

The *Dune* movies, however, took this collaborative approach to new heights. "It was a score which consisted entirely of virtuosos! And because they're all in the room or they're all connected, everybody knows the drill. Everybody knows what we're going for. We have the luxury of going, 'Hang on, we get to invent a world here.' A sonic world. It's wonderful, because once you get everybody into the room, there's no stopping them."

Nor, it seems, is there any stopping Hollywood's most daring composer.

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TalkBack

Taigen Kawabe

WILLIAM STOKES

Japanese artist and producer Taigen Kawabe is best known as the frontman and bass guitarist of boundary-destroying psych-rock band Bo Ningen, whose repertoire ranges from experimental soundscapes with hushed spoken word in English, to guitar-driven chaos and screaming in Japanese. Kawabe has numerous other strings to his bow, not least his solo project Ill Japonia, a rap project which flirts with lo-fi hip-hop and trap music, and Mainliner, his project with Makoto Kawabata of vaunted psych band Acid Mothers Temple.

At the moment I can't stop listening to

So, it's actually AI-generated music. There's a site called Suno AI, which I started subscribing to last month. But I think you can still generate maybe five or six songs per day for free. How it works is that you type the lyrics — I mean, it can generate the lyrics as well, but me and my producer friend Foodman prefer to put our own lyrics. Then you choose the genre, but you can mix any genres, or you can make up the genre as well! Foodman has made great AI music: for example, for the style he just put, like, "vinyl jazz". And it came out with some sort of weird, retro '70s Japanese music, or something.

It can be really catchy, and sometimes really unpredictable as well. You can ask it to make a structure, but I prefer to just let it do it! So it'll be more surprising. There can be weird lyrical timings. Sometimes, if I type in Japanese it'll translate it into some unknown language. It makes great errors. I mean, maybe after a couple of years they'll stop making errors.

It's interesting, from a musician's point of view. Creators are a little bit scared, wondering if our work is going to be taken over by AI In the way that's happened to the illustration world. But for us, we know the struggle of making music. How hard it is to get that bit of yourself into the sound of what you're making. AI-generated music is almost like summoning the dead! Like summoning a devil — or an angel...

The project I'm most proud of

Maybe the latest one: the remix I made as Ill Japonia for my friend's band, Black Delta Movement. And it's kind of connected to how I started making music. I've never been good at making the concept, deciding when to release, blah, blah, blah. So a remix is great because it draws that line. And it's got a new source, new layers, which I probably could not produce by myself. So it's always kind of like finding a new voice. With each remix I put some challenge in place, depending on who I collaborate with. I wanted to put in some more psy-trance elements in the second part of this remix, which I've never tried to make before. In the last couple of years I've been to raves where I've experienced trance and psy-trance in a totally different way. I was in Japan for like nine months over Covid, and there were these secret parties. So those experiences have definitely made it into the sound! I think it shows the latest me, as both a musician and as a producer.

The first thing I look for in a studio

That's a good question! I think the first thing I look at, but also the first thing I'll ask about if I don't know what it is, is the monitor system. The speakers and the amplifier. Because it is like an instrument. And whether I'm working as a producer or in a band, I want to know the character of the studio. The microphone and everything is the input source. That's what we're going to be feeding into the desk. But the speakers are the output source, and I'm curious about that as a musician, but also as a listener. It's about the character: of the speakers, and of the room.

The person I would consider my mentor

That's really difficult one. Maybe I'll say my friend, the producer Foodman. I really respect him as a musician. But also, he's one my best mates. We met, maybe 10 years ago. He makes experimental dance music, and we've been making music together as well. I mean, 70 or 80 percent of our talk is, like, ridiculous. But we're always exchanging great music when we find it, and also great interviews we find on on the Internet, to keep us excited. We talk about how difficult it can be to survive as a musician. We encourage each other.

This kind of links back to what I was saying about AI music. From talking with him, from making music together, I've learned how important it is to find that human error in the music. Obviously we can make glitches with a pedal, or with a plug-in or with Max MSP — some kind of programmed glitch — and I'm super into that. But after I met Foodman... I feel like he experiments with things like how his body or his brain can change his output. And that's always been my interest, when it comes to making music or even just as a human being. How to make errors, how not to predict what's going to happen. As a musician, as an improviser, as a human, I think I learned that from him. How to enjoy error.

My go-to reference track or album

I think everyone's probably got one reference track that they change, like, every couple of years. For me, the bass frequency is pretty important. And even in my room it is a little bit hard to know how it's performing. So I always check next to a track called 'Between Here & Yesterday' by a techno house artist called Sigha. That track, the bass on it, even on some larger speakers, you cannot hear that sub. So it's actually kind of a good reference to see if the sub is too much! Or to see if you don't hear the sub, you know, maybe you won't hear like under 30Hz, or

whatever. Obviously it's got hi-hats and all the high frequencies too, so you can also check those for reference. But it's the bass frequency that I like to check by it.

My top tip for a successful session
I remember reading something by Rick Rubin about the importance of feeding good food to the artists he's producing. And I do agree with that! I think the most important thing in the studio is to make everyone feel comfortable, but at the same time, to feel like they can push themselves a little bit. As humans, we need to feel comfortable in the studio. We cannot get nervous. But as a musician, I think we should find some kind of uncomfortable zone. Where you can be challenged — as a singer, for example, or as a guitarist.

I think if it's a 'proper studio', then it's about finding your comfort zone as a human but finding a challenging, uncomfortable zone as a musician. Make some kind of list of challenges: what you haven't done, what you want to try out. And if it's a band, you can make little lists for each other, maybe! But it can't be, you know, "We cannot have any arguments!"

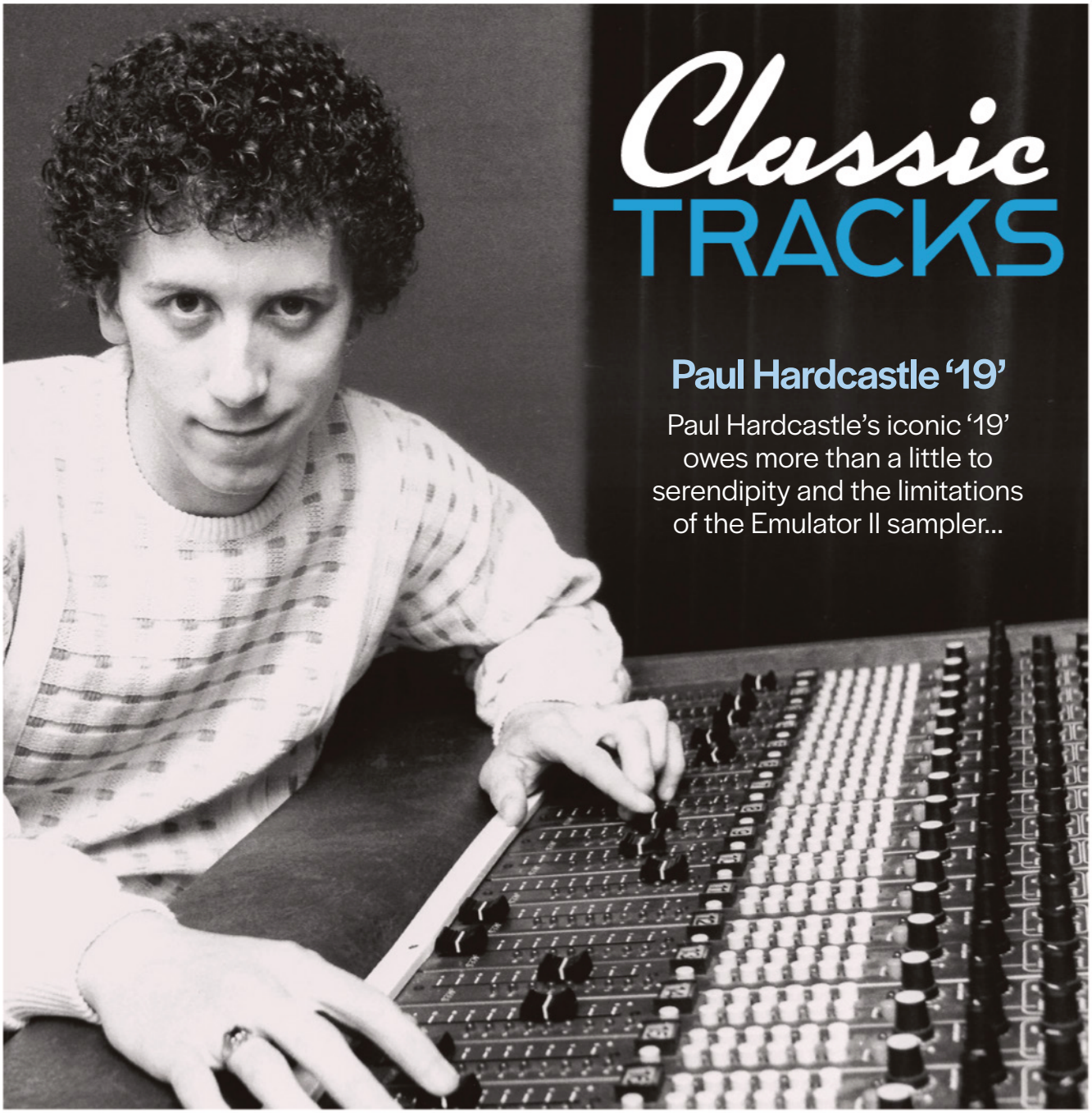
The studio session I wish I'd witnessed
I wonder, maybe one of Steve Reich's recordings? Maybe *Clapping Music*, or in fact *Music For 18 Musicians*, which is my favourite music by Steve Reich. I have no clue about how people record, like, orchestras. Or maybe Brian Eno's *Music For Airports*, that kind of thing. I could make you a list of 20 or 30 producers. Oh, actually: King Crimson. *In The Court Of The Crimson King*. Particularly the song '21st Century Schizoid Man'. That song was a huge shock for me when I heard it at maybe 16 or 17, a couple of years after I started to play the bass. My friend who was into the Beatles and into jazz showed me King Crimson in pitch black in his room, with the Windows Media Player visualiser going, if you remember that! As an experience for a teenager playing the bass, that song, those recordings... it was just really unique and amazing. I was shocked. I had never heard that kind of prog music before. I think our Japanese version got a couple of bonus tracks, maybe. And it had liner notes — sort of a biography, and also someone explaining the history and the story of the album. One CD I borrowed from my friend had liner notes about Robert Fripp, who was really inspired by Brian Eno and Steve Reich. And that's how I started to listen to Brian Eno and Steve Reich. That opened my mind.

The producer I'd most like to work with
I think I'm going to say Laraaji. He is quite old now — he's 81 years old. I listened to some spiritual New Age music while I was half asleep, and then Spotify played one of his tracks. And I was almost hypnotised, because I was half asleep. I do have sleeping paralysis quite often. I had it a bit more when I was younger. But anyway, that music almost hypnotised me. I was like, "Wow, who is this artist?" Then I found out it was Laraaji. It was music he had made when he was 20 or 30. It was so natural, and pure. I think if I wanted to collaborate with someone, to be in a studio with someone — even though I said it's important to give yourself challenges — I'd get too nervous if it was with Steve Reich or Brian Eno, or Robert Fripp. But Laraaji... He's a living legend, but one who I wouldn't get too nervous to make music with!

The part of music creation I enjoy the most
I think I have two moments that I enjoy the most. The first is when I feel, "This is going to be something." I always try to experiment, to make something from scratch rather than try to recompose from what I remember, or what's being played in my mind. It could be just a one sound. Or it could be after 70 percent of the production is built. Because you know, sometimes you're just not sure! I love the moment when the music or the sound is going beyond my expectation, becoming something I can fully enjoy. Because, well, my attention span is really short. I easily get bored of what I listen to, but also of what I'm making as well! Yeah, so when I find something great, something more than I expected, it's great pleasure, it's some kind of hope.

The other moment is like when I'm rendering the file, there can just be that magic moment, where you love your own song. It's like giving birth or something. Sometimes you struggle to even make one minute of music. Yeah, so those two moments: the moment when you feel something, then the moment when it's complete!

The advice I'd give myself of 10 years ago
Well, I still do keep telling myself I should learn music theory! And more production techniques! But I'd say: don't worry too much. Don't get too nervous.



Classic TRACKS

Paul Hardcastle '19'

Paul Hardcastle's iconic '19' owes more than a little to serendipity and the limitations of the Emulator II sampler...

TOM DOYLE

A UK number one single in 1985 (and international hit in 16 other countries besides), Paul Hardcastle's '19' was one of the first records to showcase the sonically manipulative powers of the sampler. With its stuttering "n-n-n-n-nineteen" hook locked to a rolling electro beat, it drew the listener into a spoken-word commentary detailing the death toll of teenage American soldiers in the Vietnam War, and spotlighting the PTSD suffered by many of the veterans who returned home.

Hardcastle's inspirations for '19' included the Roland TR-808 grooves of Afrika Bambaataa & the Soulsonic Force's early '80s hip-hop tracks 'Planet Rock' and 'Looking For The Perfect Beat', and a TV documentary that he recorded one evening on a Betamax video tape, later sampling its narrator, Peter Thomas, as the lead voice on the track.

"I just happened to be looking through the paper and seeing what was on the telly later," Hardcastle remembers today. "And it said, 'Vietnam Requiem', which was the story of the young kids sent to war in Vietnam. Now, I didn't know nothing about Vietnam. I don't even know why I recorded it. I just thought, 'Wow, that's crazy. I wonder if any of this could be made into a record... telling the story, over music.'"

Recorded using an E-mu Emulator II sampler and a TEAC half-inch eight-track tape machine in the front room of his house in Leytonstone, East London and mixed at Sound Suite Studios in Camden, '19' stayed at the top of the British singles chart for five weeks. Thanks to Hardcastle

Artist: Paul Hardcastle
Track: '19'
Label: Chrysalis
Producer: Paul Hardcastle
Engineer: Alvin Clark
Year: 1985

and his record label Chrysalis releasing remixes of the track during its chart-topping run ('The Destruction Mix', 'The Final Story'), it staved off stiff competition from Duran Duran, who had to settle for number two with their James Bond theme, 'A View To A Kill'.

"Duran Duran were most probably the number one band in the world at that particular time," Hardcastle laughs. "I didn't have anything against Duran Duran.

"We turned those [*remix*] records around within days. Chrysalis were very good at that. I was working with a guy there and we would re-edit the video for *Top Of The Pops*. You had different videos for it and the record stayed there."

Eight To Nineteen

By the time '19' was released in 1985, London-born Paul Hardcastle already had a successful recording career up and running. Music was in his blood. Back when he was a kid, his American trumpeter dad, Louis, often invited his son onstage for guest turns at his jazz gigs.

"We used to play a lot of American bases," Hardcastle recalls. "My dad taught me to play the drums. I was about eight and I wasn't a great drummer. But someone that could play the drums at that age, it was a novelty. I could also play the guitar. He got me a four-string guitar and I could play sort of two or three chords on it. In the end, he said he found it hard to get shows without me."

During his teenage years, Hardcastle's passion for motorbikes took over, until he suffered a nasty accident. "I had three compound fractures," he says. "I was in traction for three-and-a-half months."

His recovery period was to last a full 18 months, during which time Hardcastle swapped a video camera he owned for a Minikorg 700S synth and began dabbling once again with music. "Three or four years before, I used to go down to Macari's in Charing Cross Road. They always had one, plugged in, and there was a slider on the front called a Traveler. And if you pressed it, it made that whoosh sound. And I thought, 'Yes! I'm Simon House out of Hawkwind.' I was a massive Hawkwind fan."

At home, Hardcastle began teaching himself chords, by overdubbing individually played notes from the »

A young Paul Hardcastle on drums with his father, Louis Hardcastle.

Paul Hardcastle still owns the Sequential Prophet-5 he used on '19'.

» monophonic synth using two cassette machines. He then responded to a music paper advert placed by a Brit funk band, Direct Drive, seeking a keyboard player. "I had started listening to club music," he says, "and I was starting to get OK doing bass synth and little lead lines.

"I joined Direct Drive and then we did a couple of demos. I was able to buy us a little Portastudio, the very first little [*Tascam*] four-track on cassette. And we started to be able to make listenable demos, rather than just us live in a basement, which sounded terrible."

Hardcastle's first experience of a professional studio, a 16-track named First Light in Penge, South London, was an eye-opener — particularly when it came to the mixing stage. "We had tried to mix, but every single fader was at the top," he laughs. "Everyone wanted to be heard more.

"In the end, the engineer saw we were really out of our depth. And he said, 'Look, why don't you just let me basically get some decent levels for you? And come back in a couple of hours?' We came back and we were like, 'Bloody hell, that sounds good!' So that was, I guess, the first insight into knowing what a producer would do."

In 1982, after two singles with Direct Drive, Hardcastle and the band's singer Derek Green broke off to form a duo. Appropriating the First Light name for their new band, they released their self-titled debut album the following year and a second album, *Daybreak*, in '84.

When the pair split not long after, Hardcastle embarked upon a solo career with a cover version of New York electronic funk duo D Train's 'You're The One For Me' — a record that had blown his mind when he heard it in a club. "I remember hearing that record," he says, "and thinking, 'What the hell?'"

Following a similar synth-funk path, Hardcastle's breakthrough as a solo artist came in 1984, with his fifth single, 'Rain Forest', which reached number two in the US Dance chart, selling more than half a million 12-inch singles. Back in London, this caught the attention of a young Chrysalis Publishing A&R man, Simon Fuller, who invited Hardcastle to come in for a meeting to let him and his colleagues hear any other material he'd been working on.

One of the tracks Hardcastle played them was an early version of '19'. It baffled the Chrysalis personnel, but so enthused Fuller that he offered to quit his job to manage Hardcastle. "I said to Simon, 'What does a manager do?' And he said, 'I suppose I just take all the crap for you really.' I said, 'All right then, let's do it.' Two to three months later, we were on top of the world."

Emulator Influence

The distinctive beats on '19' (as with 'Rain Forest') had been created using a Roland TR-808 that Hardcastle had hired for a few days. "I just put beats down [*to tape*] because I knew that we had to take the machine back. I did four or five. That's where 'Rain Forest' came from, and with '19', I had a similar type of beat."

Hardcastle's synth workhorse at the time was the Sequential Prophet-5. "Always the Prophet," he says. "The first time I ever saw and heard a Prophet-5 was Andy Stennett of Freeez on *Top Of The Pops* doing 'Southern Freeez' in 1981. It went into this really nice, sort of pad type string sound and I thought, 'God, that sounds great.' Until I saw the price (laughs). I've still got my original one."

The early version of '19' that Hardcastle played Simon Fuller already featured its signature orchestral stabs, created using the Loop Editing System of the AMS DMX 15-80 Digital Delay Line. "This was before me having a sampler," Hardcastle points out. "And so we were using an AMS, that you could put sort of like a second into, for the orchestra stab. So when you hear 'dat-dat-dat'... that's just made pressing that button."

Hardcastle's eureka moment with '19' followed his acquisition of the 8-bit Emulator II sampler. "Basically, I just felt there was something missing in '19'," he says. "Simon managed to get me the money from Chrysalis to be able to buy an Emulator. It was one of the first ones that had come out. I remember this big box arriving at my house where I was living with my parents and my girlfriend in Leytonstone, and thinking, 'Wow'. The sounds were obviously all on [*five-inch*] floppy disk and the first sound I heard was the 'oohs' and 'aahs' [*factory vocal patch*]. I was like, 'Oh my God.'"

Hardcastle proceeded to record parts of the narrative from the Vietnam Requiem documentary from his Betamax video onto his TEAC eight-track, before relaying the best sections into the Emulator II. He started with the lines, "In World War II, the average age of the combat soldier was 26. In Vietnam, he was 19." "I thought, 'That sounds good against this beat,'" he

recalls. “And I just dropped other bits into it, y’know… ‘None of them received a hero’s welcome.’.”

The limited amount of sampling time available in the Emulator II — a whole 17 seconds in total, which could be divided up into one- or two-second snippets or used in one splurge — was to actually aid Hardcastle creatively when it came to inputting narrative snippets from the documentary into it.

“That’s the reason why you only get, ‘In Vietnam he was 19,’” he explains. “That did me a bloody good favour, because you can imagine if I’d have had all this sampling time, that little chorus would have never happened. I would’ve put 20 minutes’ worth of sampling in it (laughs). I mean, I could fill up a 24-track tape reel quicker than you could down half a pint.”

Once he’d carefully chosen which bits of voiceover that he wanted to sample, Hardcastle then began to manipulate them. “I thought, ‘Right, let’s start using this sampler now. In Vietnam, he was 19’… click, that was the end of the sample time. And I just hit the key — ‘in, in, in, in Vietnam.’ Then I cut it up to the ‘19’ and hit the key and I just remember thinking, ‘Fuck me, that works.’ ‘N-n-n-n-nineteen.’

“So, I put it down and then I went out with friends. I came back and I thought, I’ll just have a quick listen to that mad thing I was working on. ‘Hey, that sounds pretty good.’ Then all of a sudden it was like, ‘N-n-n-n-nineteen’ and I went, ‘Shit.’ You know when you do something and you forget about it, and it sounds great again after? I just amazed myself: Oh, wow, this is a hook.”

Running Out Of Tracks

Hardcastle then added other key sounds to the ‘19’ multitrack, including a synthesized ‘whu-ow’ on the Prophet-5 that became a sonic feature of the track. “That sound I’d used on ‘Rain Forest’,” he explains. “Like a piano player [*runs their hand*] from the top to the bottom of the keyboard, it was basically me with my left hand, sort of going up, dow-wwwah, and then with my other hand [*opening up the filter*]. You could actually go and make that now in two seconds if you wanted to. Any old synth, just get a sound that’s got tons of resonance. Just turn the resonance up to full and you’ll have that sound.

“I also had just got a [*Yamaha*] DX7. There was the remix, ‘The Destruction Mix’, that had sort of like a marimba-y sounding thing. Because ‘19’ was very dark, I just played some minor chords to it. ‘Log Drum’ was most probably what I used, played higher up the keyboard, and I just remember going ‘dt-dt-dt’ and it worked perfectly. There was no sequences, obviously.”

Melodically, ‘19’ relied mostly on its synth hook lines, until Hardcastle decided to bring in a session singer, Janice Hoyte, to perform an additional top line. “I just said to Simon [*Fuller*], ‘I need someone. I think it would be a good idea to break it up and have some sort of female chorus type of thing on it to give it more colour.’ So he got me a vocalist. And I had just written these few lines, y’know, ‘All those who remember the war/They won’t forget what they’ve seen…’”

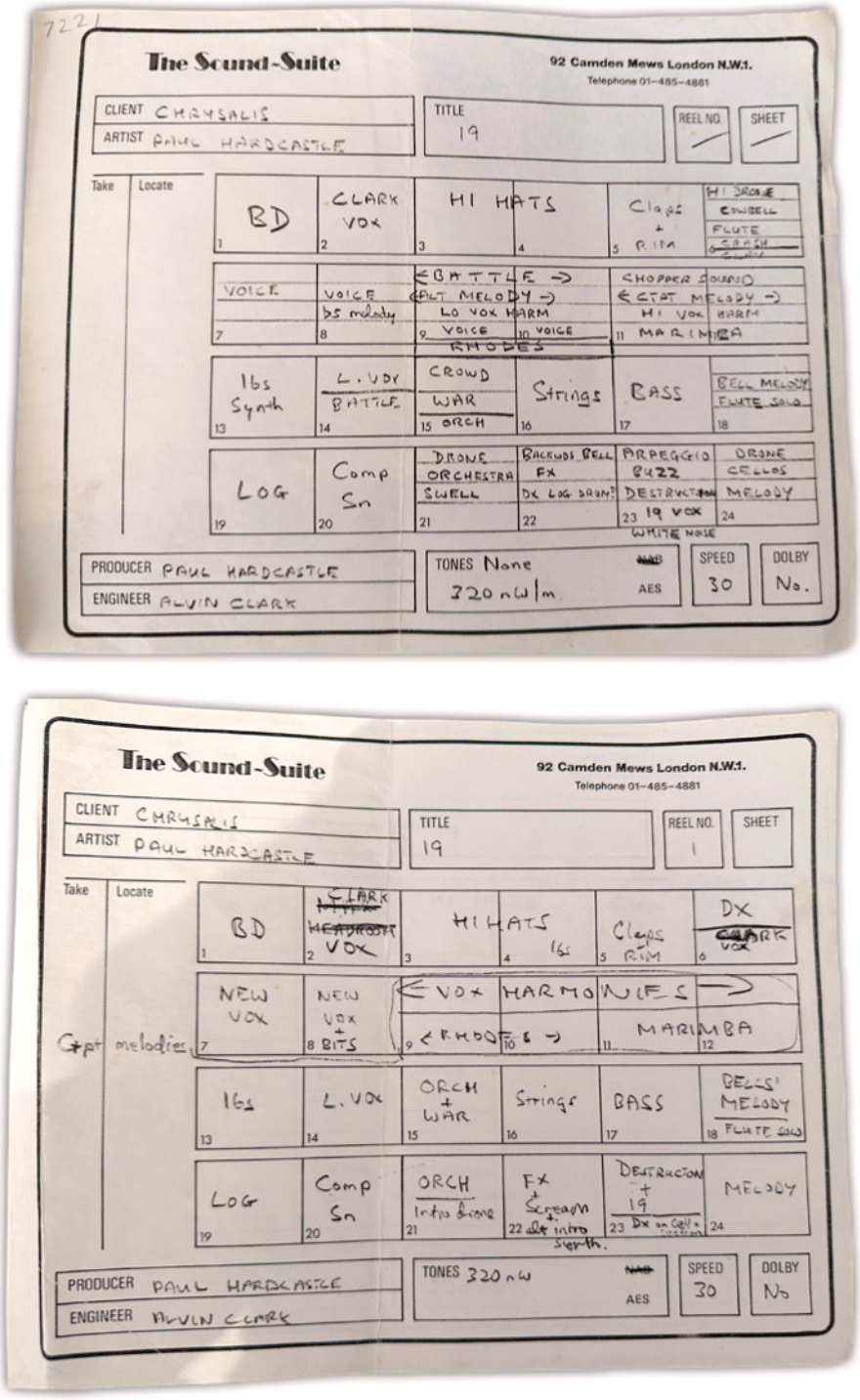
Hoyte was recorded at Sound Suite Studios in Camden, where Hardcastle also mixed the original version of ‘19’ (and completed its later remixes). “We took the eight-track into Sound Suite, bounced it over. And we had Dolby then which took a lot of the noise down. Because we hadn’t done the vocals by then, it was raw… just the drums, the bass synth, the chords, and the noises.

“I was to and fro to the studio, y’know. I’d take a mix home and I’d think, ‘Hang on, let me just bounce what we’ve done from 24-track back to the eight and add a few more bits that I want to do.’ Then, I’d take it back and put those bits onto the 24-track as well. We ran out of tracks in the end because I’d done so many remixes of it.”

Part of the international success of ‘19’ was down to the fact that Hardcastle and Fuller shrewdly employed newscasters in different territories to translate the spoken word parts of the track into French, Spanish, German and Japanese.

“The French one and the Spanish one didn’t work, in my eyes,” he admits. “I think the German version really, really stands up because of the way their language is very stark.” Later, given how many remixes he ended up creating of ‘19’, Hardcastle wished he’d made copies of the master tape.

“What we should have done really was had the master copied 10 times,” »



Track sheets for '19'.

» he says. "The multitrack, there's not a spare bit of silence on it."

Remixer

Following the success of '19', Paul Hardcastle became much in demand as a remixer for the major record companies, often those keen to add the same sampler tricks to hits from their back catalogues and give them an '80s makeover. Hardcastle's first commission was from Island Records to rework Jamaican reggae/funk band Third World's 1978 hit, 'Now That We Found Love'.

"I really did go out on an edge on that one," he recalls. "They said, 'We want you to make it really, really different.' And I suppose there were expecting the 'n-n-n-n' and all that. Well, they got a little bit of that...

"But I actually asked them to copy the 24-track and wipe it, basically. All I wanted was just the vocal and the hi-hat left in it. And I just replayed everything. I remember playing along with a Linn Drum to the hi-hat, just to keep in time. It took a couple of goes because there was no [*audio*] quantise in those days, y'know (laughs).

"I just started redoing the drums, put different chords on it, put a lot of keyboards on it, bass synth.

"I took it into Island Records... and I'm not joking, they were all dancing around the bloody office. I came out on a high, thinking, You must be doing something right in this business, now, Paul. Then the offers all came flooding in. And I did a year's worth of remixes for totally different people from Ian Dury & the Blockheads to Five Star to Phil Lynott. I mean, I was the golden boy at that time remixing everything."

At a certain point, Hardcastle made the decision to have a break from remixing to concentrate once again on his own records. Then, one offer came in that he couldn't turn down: the chance to rework D Train's 'You're The One For Me', the record that had inspired him at the start of the decade. "I went (instantly), 'I'll do it,'" he laughs. "I spoke to [*singer*] James Williams on the phone. I'm thinking, Jesus Christ... I'm about to fuck around with this guy's track and he's like a hero.

"I changed quite a lot about it," he says of the '85 version, which reached number 15 in the UK. "Put a solo on it. I always used to add my little sort of Clavinet sound from the Prophet-5. That was the first time I ever did *Top Of The Pops*. It was just a fantastic time in my life."

Legacy

After '19', Paul Hardcastle had a string of other UK chart hits. The initial pressure for a follow-up single resulted in the high concept 'Just For Money', a track themed around a heist, featuring the voices of Bob Hoskins and Laurence Olivier (which unfortunately only made it to number 19, despite its lavish £200,000 video).

Hardcastle fared better commercially in 1986 with his number eight hit, 'Don't Waste My Time' (featuring singer Carol Kenyon) and with 'The Wizard', which became the *Top Of The Pops* theme tune for five years between '86 and '91. The latter track made heavy use of Hardcastle's new $16,000 Synclavier

sampler/workstation, paid for with the royalties from '19'.

Meanwhile the commercial appeal and cultural impact of '19' endured. In 1988, the track inspired a spin-off game for the Commodore 64, titled *19 Part One: Boot Camp*. Much later, in 2019, the radio station Absolute '80s created a remixing app for iPad that allowed users to rebalance and effect the original stems.

In 2015, for its 30th anniversary, Hardcastle released an entire album of new and old remixes of '19'. "I did 12 new ones, basically," he says. "I thought, 'Y'know what? Let's get the old girl out one last time.'" Among the new remixes, the 'Cryogenic Freeze Remix' expanded on the electro foundations of the original, while the 'PTSD Mix' was much tougher and more modern-sounding, verging on drill.

"We gave the profits to a charity called Talking2Minds that helps soldiers with PTSD," Hardcastle says of the latter mix.

"I had tons and tons of letters, y'know, I'm talking 10,000, from American vets, back in the '80s. They just didn't understand why a young British guy was taking on their fight... because they had to fight to get recognised. And as the track says, 'None of them received a hero's welcome.' They used '19' when they marched through Washington on their protests. I've had at least eight or 10 letters that say, 'Dear Paul, thank you for saving my life.' What more could you want?"

Elsewhere on the remix album, the downtempo 'NUA Remix' featured Marvin Gaye's iconic vocal hooks from 'What's Going On' alongside '19's stuttering interjections. The suggestion that the two tracks could work together in fact came from Hardcastle's father, Louis.

"'19' was really all about smashing it in your face," Paul stresses. "But you listen to the one where it's slow, and it works in a totally different way. Because it makes it very, very sad, having Marvin in it: 'Brother, brother, there's far too many of you dying,' and then it goes, 'the average age was 19'. It was just like, Woah. I felt like I really created something special."

The people who administer Marvin Gaye's estate agreed and gave Hardcastle clearance to go ahead with the release of the track. "That took a year," he points out. "I remember getting the email from the woman that was doing the clearances for me: 'Dear Paul, good news. We've finally got clearance for you.' It's like watching your team win the FA Cup when you're told you can use Marvin Gaye, and it's legal..."

The legacy of '19' also extended to the name of Simon Fuller's 19 Entertainment management company, which famously launched the career of the Spice Girls. Hardcastle also wrote the score for their 1997 film, *Spice World*.

In terms of his own subsequent career, Hardcastle has gone on to release a series of smooth jazz albums, both under his own surname and as the Jazzmasters, many of which have made the Top 5 in the US Jazz chart.

"It was a career change, obviously," he says. "And the career change really turned into going back into sort of like the 'Rain Forest' type of stuff."

Nonetheless, Paul Hardcastle is still clearly very proud of '19', his first big success, and the track that made his name. "It's the same with both me and Simon Fuller," he says. "Y'know, we talk sometimes, and we go, 'What if we'd never done this?'

"People say, 'Do you ever get tired of hearing "19"?' I say, 'No'. Well, put it this way, I don't ever play it myself. But when it comes on the radio or something like that, I sort of smile."

Ultimately, Hardcastle reckons that, sampling novelty aside, it's the anti-war message of '19' that's ensured that it's become a classic track.

"It was very different," he says. "It was very truthful. It was stark, it was in your face. And I think a lot of people appreciated the fact that I had the guts to basically do that.

"Another point, which I find very important is in the US you can't get a beer 'til you're 21. And you can go out at 18, 19, and be dropped off in a fucking jungle. Basically, I call it a musical documentary. It was just letting people know, in a different sort of way, what happened."

Paul Hardcastle's box set Nineteen And Beyond 1984-1988 is out now.

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INSIDE TRACK

Daniel Nigro ▪ Olivia Rodrigo

Guts has cemented Olivia Rodrigo's status as a major star. Her chief songwriting and production partner is Daniel Nigro.

PAUL TINGEN

In January 2021, Olivia Rodrigo's 'Drivers License' was a global hit, going three times platinum in the UK and six in the US. Rodrigo, only 17 when the song was released, proved anything but a one-hit wonder, because several more multi-platinum songs followed that same year, 'Déjà Vu', 'Good 4 U', 'Traitor' and 'Brutal'. They all appeared on her debut album *Sour*, which also went multi-platinum. The album and breakthrough hit won her three Grammy Awards, plus four further nominations.

These spectacular achievements were facilitated by Daniel Nigro, who co-wrote eight of *Sour*'s 11 songs, and produced the whole of the album. Nigro had already enjoyed hits with Sky Ferreira, Freya Ridings, Caroline Polacheck and Conan Gray, but the success of *Sour* took his career to an entirely different level. And when Rodrigo and Nigro started work on the singer's "difficult second album", they did indeed find the going difficult.

"It was a hard process," admits Nigro. "The first half of making *Guts* was difficult because there's that constant voice in your head saying, 'It's not good enough!' or 'How are you going to beat this?' There was a lot of pressure as we had *Sour* in the back of our minds, a body of work that was finished, mixed and mastered. And next you're working on songs in your studio, listening back and going, 'Well, that doesn't feel as magical or special.'

"This was in the beginning. But with real hard work and real dedication, the special songs started to show themselves. If you keep showing up every day, some good things happen. I can remember days in the studio where I didn't know if I could get through it, having been up all night with my baby and me being exhausted. Some days you fight through and nothing good happens. Some days you fight through, and then, all of a sudden, you're like, 'Wait, did we just come up with something that we like?' This started to happen more and more."

The hard work paid off, and Rodrigo and Nigro managed to deliver another hit album. Released in September 2023, *Guts* reached number one in dozens of countries, contained the multi-platinum hit single 'Vampire', and received six Grammy Award nominations, including Album of the Year and Best Pop Vocal Album.

Sour And Sweet

Olivia Rodrigo's success did not come entirely out of the blue. She had enjoyed lead roles as an actress in two Disney+ series, *Bizaardvark* (2016–2019) and *High School Musical: The Series* (2019–2022) and, as her character in the latter series, had enjoyed a hit with the self-penned song 'All I Want' in 2019. Despite this, she was unknown to Nigro when he first encountered her in 2020.

"A friend in the industry was curious to know if I had been working with Olivia, because she was following me on Instagram, but I had never heard of her before. Most of the time when somebody asks you to listen to something, you go 'OK, whatever,' and you move on. But when I listened to her song 'Happier', which had a 30 second video, I got the chills. I remember thinking, 'What a clever concept, that's really good.' She had so much conviction in her voice and so much passion. As real as it gets. I reached out to her, and after many months, because of the pandemic, we ended up finally meeting, and 'Happier' became the first song we worked on together.

'Happier' was later included on *Sour*. Nigro, from his side, was at a point in his career when he was seriously starting to get into production, after having been an artist with the indie rock band As Tall As Lions, and a songwriter for the likes of Sky Ferriera, Kylie Minogue, Billy Idol, Lewis Capaldi and many others. By the time he started working with Rodrigo, Nigro had just enjoyed his first production successes, also co-writing and playing many of the instruments, with Freya Ridings' song 'Castles' and Conan Gray's album *Kid Crow* (2020), which had gone to number 5 in the US.

Sowing The Seeds

Nigro wrote and recorded most of *Sour* and *Guts* at his own Amusement Studios (see box). Nigro describes their process: "Often Olivia comes in with a seed, like a chorus or something. Some days she comes in with just a lyric concept. She could be like, 'I want to write a song with this title.' And then we go from there. The great thing about working with Olivia is that everything is done from scratch. It's usually just her and me with a guitar or a piano. We sit in the room, talk about it, and try out different ideas, and see what happens.

"The song is the most important part, and to be honest, most of the time we initially don't think about the production, or what the sonic landscape of a song should be. It's always about the song, having a great chorus that we both really like. From there we can dress it up however we want. Sometimes we write a song and I go 'OK, cool. I know what this is supposed to feel like,' and then I produce it and it works great, but sometimes she might question the arrangement or production, and I have to go back and try again, and again. It can take a few times for a song to find its identity.

"Olivia has strong opinions on the production, and doing it is a team effort. It's a happy accident that my skill set happens to match her sonic tastes. She loves live guitars and live drums. At the same time, there's tons of programming. »

The drum kit at Amusement Studios is permanently set up and miked with a pair of Coles 4038s as overheads.

Olivia Rodrigo's debut album *Sour* was a huge global hit.

» On the first record all but three songs were predominantly programmed. But for *Guts*, Olivia really wanted things to be more organic, and have more live instruments. For example, 'Bad Idea Right?' has live drums, but the kick and snare are programmed. With most songs I program the drums first, and then a drummer enhances this by pulling and pushing.

"The drums on 'Ballad Of A Homeschooled Girl' are entirely live, apart from a snare in the background that I added later, to get it to sound exactly like I wanted to. Olivia likes the push and pull of live instruments. I can create that in the box as well, but massaging the parts takes a long time. It's days of work. People call me 'Nudge Nigro', because half the time I'm messing around with the feeling of programmed parts, to add humanness by nudging things in the timeline. It's surprising how much it can change a recording!"

Vampire Beats

The sessions for *Guts* also made use of EastWest and Skylight Studios in Los Angeles, and Electric Lady Studios in New York. A large number of additional musicians and engineers were involved, including guitarist Sam Stewart, multi-instrumentalist Ryan Linvill, drummer Sterling Laws, pianist Benjamin Romans, violinist Paul Cartwright, as well as engineers like Dave Schiffman and Dan Viafore. Many of them were in action on 'Vampire', the lead single and main hit from *Guts*, which Nigro uses to illustrate his and Rodrigo's process in more detail.

"Olivia wrote the original seed for that song, and brought it to me when we were in Electric Lady. It had a different verse, but the chorus was pretty similar, with different lyrics in places. The basic concept was there and the 'Bloodsucker, famefucker' hook was there. I thought that was incredible. We decided to rewrite the verses, and then rewrote some lines in the chorus, to make them really strong. We spent an entire day on this, really honing in on the lyrics, and making every line feel like it meant something.

"We got two verses and two choruses, and we recorded a demo, using an upright piano and an iPhone. Almost every song that we made on *Guts* started as an iPhone demo! When we got back to LA we were really excited about the idea. It was the first song I was going to produce, and I had Sterling [*Laws*] come over, and recorded a drum beat that I thought was appropriate for the song, which was a half-time groove. I added a couple more instruments, and when Olivia came over the next day, I played it for her. To my surprise, she really didn't like it. She said, 'It shouldn't be that, it should be much more intense.'

"It had not been my instinct, but I realised that she wanted the song in double time. So I programmed double-time drums. I go through my drum sample library and pick out kicks and snares, weird little clinks and clacks, and extra cymbals, to make it all feel bigger, and put them in the timeline. She liked the double time, and from there we started to sort out the rest of the song. We wrote the bridge a week later. Sterling re-recorded the drums, and Paul came in and laid down 100 tracks of strings. I used a Mellotron in the second verse, and added organ sounds from the [*Roland*] Juno 60, one of my favourites. The song really started to take shape."

Three Become One

Nigro and Rodrigo assumed that they were on a home run with the first song for the album, but there were still some

Finding A Voice

Growing up on Long Island in the '90s, Daniel Nigro took guitar, piano and singing lessons as a teenager. He then co-founded As Tall As Lions in 2001 with several high school friends, and Nigro as the lead singer. The band released three EPs and three albums between 2002 and 2009. During his time it gradually dawned on Nigro that the touring lifestyle was not for him. After visiting a songwriter-producer friend, Justin Raisen, he decided to follow the same career and moved permanently to LA a year later.

"My intention was to get into the jingle game, make commercial music and potentially write some songs. I got a real sense of fulfilment out of being in the studio, and creating something that didn't have the pressure of being the artist. I enjoyed being creatively free, and being able to put on a different hat every day. I'd get briefs like, 'Make a thing with shiny guitars and an upbeat drum beat.' I would never make music like that instinctually."

Nigro worked in a restaurant for four days a week, to make money, until a jingle he wrote for McDonalds kept him afloat for a couple of years. In addition, his friend Justin Raisen introduced him to Ariel Rechtshaid, who already had a track record with Plain White T's and Glasser, and would go on to work with Haim, Beyoncé, Madonna and many others.

"Ariel was creating the Heavy Duty publishing and production company, and Justin and I got a lot of work via him, writing songs and jingles. Justin and I would create what I call 'seeds', where we would write unfinished ideas, like a verse and a chorus, and we'd send them to Ariel. If he liked a seed, he would complete and produce it, and make it sound like his sonic world, and then he'd bring that to the artists that he worked with.

"I did that for the first three or four years that I lived in LA. But then Ariel became so popular as a producer that it became hard for me to get time with him. So I also worked with other producers, and found myself going into rooms to write songs, without having a creative say over how the songs were finished. It dawned on me that when I followed what I thought was working for other people, I always ended up not being happy. So I started to develop my skill set, as a producer. I realised that I'm really good as a creative partner for someone who knows what they want to say. I'm not the one to come up with the lyrical concepts of the songs, but when it comes to what the song should feel like and the melodies, I'm really good.

"I gravitated to people who have a strong identity as an artist, and once I found a couple of artists that I liked, I put a lot of energy into them. These were artists like Caroline Polachek, Freya Ridings and Conan Gray. I could see what makes these artists incredible, and I want to help them get to that next stage. That's when I started to see myself as actually having success in music. Working with Olivia Rodrigo pushed the whole thing to a completely different level."

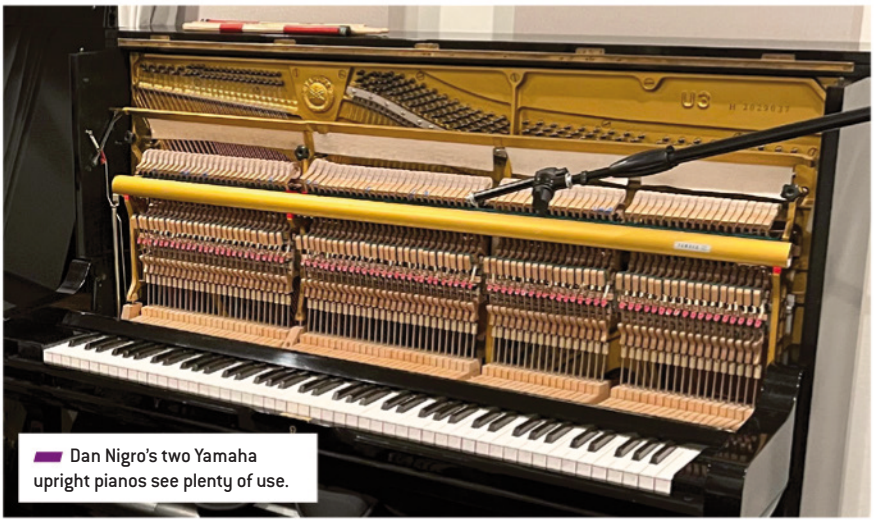
twists and turns to come. “The funny thing was it was becoming a miniature opus. It was at the end of January 2023, and very few people had heard new music. Her management was very eager to hear something new, but I was not keen to send things out that were unfinished, because people are going to listen to it a couple times, and if they don’t like it, you have a problem, and if they like it, they’ll get demoitis.

“We did in the end invite some people over to the studio to listen to some music, and we were really excited about ‘Vampire’. It was a sketch, but all the parts were there, it just wasn’t finished. We played the song and they were like ‘OK... it sounds like three songs in one,’ and we were like ‘Yeah! That’s what we’re going for!’ And they were like ‘OK, but it sounds like it’s three songs in one.’ I think everyone was a little scared, wondering whether we knew what we were doing. It was a sign that I needed to work more on the song, and massage the parts so they really flowed together nicely.

“I finally managed to come up with a new demo, which we felt was close to finished, and we sent it to everybody. The reactions this time were very excited, but then Olivia and I couldn’t figure out the tempo of the introduction. The song gradually gets faster and faster as it goes on, and we couldn’t agree on the tempo of the first verse and chorus. She kept wanting it slower and slower, and I felt like it was too slow where it didn’t have enough urgency in the vocal. We kept making version after version of the song, with the bpm going back and forth around 135/133/132. This isn’t an issue when you’re programming, but because we wanted it to feel natural, we had to re-record it. So we just kept on re-recording the first verse and chorus, from scratch.

“Finally, we agreed on a tempo. She was happy, I was happy. And I said, ‘I’m going to go Sunset Sound to record the piano in the intro, because I really love the way their grand piano sounds. And I think that the song needs to be on a grand piano.’ I loved the upright piano at Electric Lady, but it didn’t work for this song, and I then re-recorded the piano in my studio, but we felt it sounded too clean, with not enough body. It really needed a grand piano.

“We all went to EastWest, and recorded the grand piano, and re-recorded Olivia’s vocal. We were all very excited. And »



Dan Nigro’s two Yamaha upright pianos see plenty of use.

» then she said, 'It's too fast,' and I had to slow down the first verse by 1bpm. We couldn't go back to Sunset Sound, so I went into Pro Tools and somehow figured out a way to slow it down and not make it sound as if there were any artefacts. I did it by hand, moving the transients in the audio. The amount of nudging I had to do was incredible. Luckily, the tempo of the first chorus and the rest of the song remained the same."

Finishing Touches

Everything on 'Vampire' other than the grand piano was recorded at Amusement Studio. It's also where Nigro does his post-production, which ranges from vocal production to rough mixing. "For the vocals, with Olivia sometimes the demo take is really incredible and we just use that. Sometimes we tweak the song lyrically and melodically, and once that's sorted out we'll go in and record the vocals properly. We definitely worked hard on 'Vampire', as we were both very particular about how we wanted the vocals to build through the first verse and chorus. It involved a lot of trial and error, having her sing it and then listening back. We work together building up the vocal performance, and we then comp things together.

"During vocal recording and rough mixing I use the same plug-ins for every singer that I work with. Other things are to taste. I always throw on a Waves Q-series parametric EQ plug-in. I usually cut off low end, and then I put on a UAD 1176. So I'm double compressing the vocal — I have the initial vocal going though my silverface outboard compressor, and then there's another one in the box.

"If the song needs some grit, like with 'Bad Idea Right?', 'Ballad Of A Homeschooled Girl', 'Get Him Back!' or 'Pretty Isn't Pretty', I'll throw a SoundToys Decapitator on right after the EQ, before the compressor. I use the Decapitator for a little bit of saturation, because it makes the vocal cut through a little more. After the compressor I have a de-esser and then a reverb. The reverb is always the flavour

Amusement Arcade

Amusement Studios was originally a home studio. Dan Nigro and his family moved out to a different place two years ago, and the studio has since expanded to fill the entire house. Nigro is a Pro Tools user and has two Universal Audio Apollo audio interfaces. He has had his ADAM A7 monitors, which were used to make *Sour*, since 2011, while a pair of ProAc SM100s were his main speakers when making *Guts*. Because Nigro plays many instruments, the studio contains an impressive amount of equipment!

"My main mic pres are two stereo BAE 1073s. For compression I have a couple of Urei 1176s, one silverface and an original Rev A 'blue stripe'. I got the latter about two years ago, after we made *Sour*. I used the silverface on *Sour* and *Guts*. I tried the 'blue stripe', but it doesn't hit as hard. It's really good for when you have an open, delicate song, but when you're doing an intense vocal I find that the silverface works better.

"Next to my outboard I have a synth station, with an original Korg MS20, an original Moog Model D, a Roland Juno 60 and a Juno 106, a Mellotron, and an ARP Solina string ensemble. I feel that the low end on hardware synths sounds nicer, but maybe I'm imagining it. In any case, I also use many soft synths, including the Arturia stuff. Their Mellotron and Korg MS20 are great, as are their M1, Polysix, and so on. I love all the Korg stuff. I also love using Keyscape, particularly their Rhodes and the Wurlitzers. The Wurlitzer on 'Déjà-Vu' came from Keyscape.

"I have a Yamaha U3 upright piano from 2010, which I used on the song 'Logical' on *Guts*. But on 'Driver's License', 'Happier', 'Teenage Dream' and 'Making The Bed', I used my 1971 Yamaha U3. I have a bunch of great guitars, including my Dad's guitar from he was younger, which is a Guild D25, from the early '70s. It is at the heart of my acoustic guitar sound.

"I stereo record the guitar with one Bock Soundelux U99 below the body, and one around the 12th fret or so, and then I'll record it again in the same way, and pan the signals half way and wide, for a super-wide guitar sound that's washy with a lot of spread. It's a sound that I go for often. For instance, 'Enough For You' on *Sour* was recorded like this. But if I'm quickly writing a song, I'll just put up one microphone. Sometimes a demo-ish sound is cool, like 'Favourite Crime' has a mono guitar. And the song 'Can't Catch Me Now' for the soundtrack album *The Hunger Games: The Ballad Of Songbirds & Snakes [2023]* has a single, double-tracked U99.

"I record vocals with the U99. I also have a Telefunken ELA M 251, but on some of the female artists that I work with, including Olivia, I find it a little too bright. I prefer the Soundelux. I record all vocals three feet away from where I'm sitting, which makes it easier for me to coach whoever I'm working with.

"My two Apollos are linked, and the drums run through one of them. I have an old '60s Rogers kit in a separate room, which is set up at all times, with Coles overheads, and a U99 room mic. There's a Soundelux IFET7 on the kick, a Shure SM57 on the snare, and a [*Neumann*] KM184 on the hi-hat. But to be honest, most of the drum sound in my studio comes from the Coles overheads and the room mic. I don't ever get really great drum sounds from the close mics at my studio.

"For the most part I play the instruments at my place. Olivia plays piano and guitar great, but it just is quicker if I play and mic it up myself. It allows me to knock the parts out real fast. I usually engineer everything myself, unless we go to another studio. I'm not good enough to go to a different studio and know how to get everything going. In my own studio everything is setup and ready to sound the way I want things to sound."

Amusement Studios is no longer a 'home studio', as Dan Nigro and his family have moved out!

Valhalla's VintageVerb is a favourite plug-in and was used extensively on 'Vampire'.

in the song. My favourite is the Valhalla VintageVerb. It has many different settings and it's so easy to manipulate the mix and the decay. It makes it really easy and really quick to throw it on.

"The mixers I work with are Serban [*Ghenea*], Spike Stent, and my friend Mitch McCarthy. I send them my Pro Tools sessions, so they can start their mixes exactly from where I left off. The levels in my rough mixes are pretty close to where they're going to be. And then someone like Serban comes in and really enhances it. You get the song back with a lot more width, the low end hits better, the drums hit harder, and so on. The levels tend to be pretty close to what I sent off, but he's really great at improving the sonics.

"Olivia is also involved in the mix feedback, and sometimes there are lots of notes and sometimes not many. Serban did a really good job with 'Vampire'. That song did not take much back and forth. The main things we tried to fix with Serban was getting the drums in the right position, getting them to hit well, and also how much low end to have, because Serban adds a lot of low end, a lot of the time.

"Serban is one of the few mixers who when you call him, and tell him something is too much, will say, 'No, it's not.' I'll be like, 'There's too much low end,' and he'll say 'The low end is good!' and I go 'No, I don't like it,' and he goes 'No, it's good.' The next day I may say, 'I listened to it more and I think I'm used to it now,' or, 'No, I still feel the same way.'"

Gut Feelings

The hard work Nigro and Rodrigo put into *Guts* paid off spectacularly, not only in commercial terms but with six Grammy Award nominations for Rodrigo and three for Nigro. Other artists and producers triumphed on the night, but it demonstrates the extent to which *Guts* has established the singer as an enduring artist. "Olivia and I were pretty excited about the Grammy Awards, but we also knew it was tough, because there were so many great songs and albums out this year. I think just getting nominated is already pretty awesome. Olivia and I continue to work together, so we'll see what the future brings."

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How I Got THAT SOUND

David Bottrill: Peter Gabriel 'Passion'

JOE MATERA

David Bottrill's vast credit list runs the gamut from artists such as Peter Gabriel, Smashing Pumpkins and Tool to Muse, Silverchair, Rush and many others. Asked to pick a favourite sound from that catalogue, he chooses the drone that underpins 'Passion' by Peter Gabriel, from the soundtrack to *The Last Temptation Of Christ* (1989).

"The track starts out with the drone, which was the main building block of a lot of the pieces on the soundtrack. Because it was back in the analogue tape days, I used a 24-track tape to build multiple drone sounds. I used a Prophet-5 synthesizer as one of the main sound sources. I built a low pad-like sound and put that on one track and in a specific key. Peter often liked to work in the key of B or B minor, but if the track was to be written in another key, we could just varispeed the tape and tune it to the key desired. We had a reverb unit called the Quantec Room Simulator and it was one of the first reverbs that had a freeze on it, so I would set up a long reverb sound, Peter would sing a note into it, and I'd hit freeze. Then he would sing a second note, a third or fifth, and I would open the freeze switch and then re-freeze it and keep doing it 'til we had a chord built up.

"I had an old Vox bass guitar, and I would tune it so all the strings were either B or F#, set in front of an amp and get it into a stable feedback and I recorded that on a track. Then I would do the same with my Stratocaster. I had a lot of tools to use, so I would switch to other keyboards to make different drones or use different sources for the Quantec. I also made more drones out of some samples in the Fairlight. I would build drones on all 24 tracks, all in the same key, and then took that 24-track and used it as it was, or used varispeed to tune the tape to the key desired. Then, using all the 24 faders, I would bus the output of those tracks to another tape machine and 'play' the sound, bringing up each one separately and blend different tones to build a living drone for around 10 minutes or so. We built long pieces because we didn't know how long the cues would be, so better to be too long than too short. I then had this stereo track of this moving drone that gave you this whole atmosphere for the entire length of the track."

Over The Drone

"The first melody you hear is Jon Hassell playing trumpet, using a phrase Peter wrote for the piece. He has a very definitive sound, using a lot of breath in his playing, so it has a very unique and organic feel to the tone. I think he would blow into the trumpet as well as using his embouchure to make a breathier note.

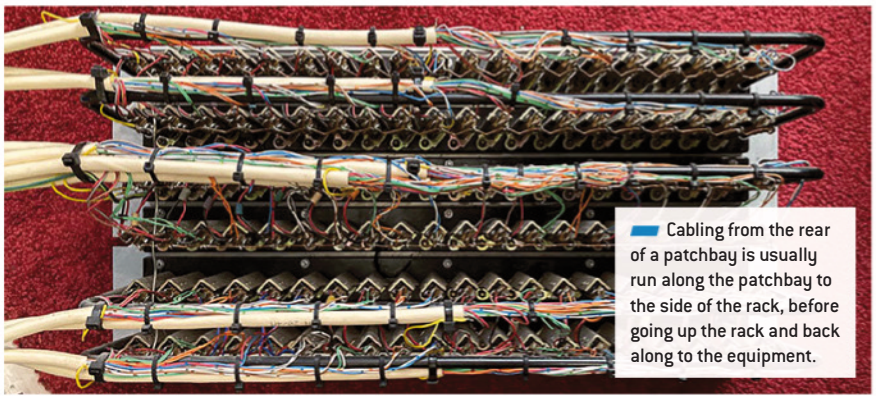
"Then Nusrat Fateh Ali Khan sang his part. I had never recorded him, and had no idea how loud or quiet he would be singing. He was sitting on the floor, and we had a U47 positioned on the floor with him. At the end of the track, I froze his last note into the Quantec and it holds his last note out for a much longer time and blends into the whole sonic texture. At the same time, L Shankar added his double violin. We would often record musicians together when improvising, so that they could feed off each other's performance. I recorded Shankar through a DI and put his signal through a Roland Dimension-D and then into a Delta Lab DL-2 with a stereo delay, often an eighth and dotted eighth note.

"To build the keyboard sound Peter was playing, I again used the Prophet-5 on a pad sound, often with the filter cutoff low and warm. I sent that off into an AMS DMX 15-80S, set on both delay and an octave up and an octave down with internal feedback, to get these octave regeneration sounds. That gets sent to an AMS RMX 16 reverb, and the whole thing sounds heavenly. It took hours to build that, so after we recorded the track, I sampled each note on the keyboard into the Fairlight CMI Series 3, so we would always have that sound to play. You can hear it all over the soundtrack."

Hear The Sound

W www.youtube.com/watch?v=czagvxCpcHY

W https://open.spotify.com/track/0P9tEFpT2IUIJi09mKRfB3



Cabling from the rear of a patchbay is usually run along the patchbay to the side of the rack, before going up the rack and back along to the equipment.

Q Should I use my patchbay upside down?

I'm wondering if it makes sense to flip my patchbays (Samson S-Patch Plus types) upside down in my rack, to put my mixer inputs on the top row and signal outputs on the bottom. This would help keep the rear cables connecting the patchbay to the mixer (above the rack) out the way, but preserve the 'listen-or-break' function in half-normal mode. Turning the patchbay upside down also flips the conventional orientation (and printed numbers) of a patchbay, so listening to the signal on the front bottom and breaking it on the front top. Is this a common reconfiguration (especially with a mixer above a patchbay) or is there a reason for the convention is the way it is?
***SOS* Forum post**

Hugh Robjohns, Technical Editor

I don't know the historic reason for the 'outputs on the top' convention, but I suspect it's related to the practicalities of wiring between the switch contacts on the old long-frame jack sockets to set up normalling schemes. Nevertheless, 'top-out, bottom-in' (as viewed from the front) *is* the convention, and all pre-wired and modular patchbays are configured in this way.

There's nothing to prevent you from mounting your own patchbays upside down but I'd strongly caution against it, as it's likely to lead to confusion if anyone uses your studio, or if you use anyone else's! You'd have to re-label everything too, as the socket numbering and switch idents will all be upside down too, of course.

Plus, while I understand that it might initially seem neater if the cables going to the mixer channels can be looped straight into the top sockets using short cables in straight vertical drops, the professional approach is to run (longer) cables horizontally across the back of the patchbay to the side of the rack, and then route them vertically up or down to the appropriate level of the rack, where they route horizontally again to connect to the equipment. This keeps things neat and well-organised at the back of the rack and, with a small loop of cable left at the edge, allows units to be removed from the rack with the cables still attached, instead of having to dive into the back of the bay to unplug everything before unmounting. Admittedly, this cabling technique is easier if you can make up your own cables rather than using pre-made looms, as you can cut each cable to the exact length required, but it's possible with looms, with a little more effort (and cable ties).

Q How can I train myself to identify vocal problems?

How do I make my ears more sensitive to inconsistencies in the vocal frequency spectrum? In theory, I seem to understand, but when comparing my mix with a reference track, I always feel that my vocal tone needs to sound clearer and more solid. Yet, I'm not sure what frequency range I need to work on.
Anon, via email

Mike Senior, mixing expert

A lot of what makes a vocal sound clear and solid in a mix isn't actually EQ: it's arrangement, dynamics, fader automation, effects, and how the backing track parts are processed. EQ is only ever part of the solution. But there are a number of tricks I use to assist me in identifying inconsistencies in the vocal spectrum.

Turning the monitor volume up loud is great for highlighting whether certain consonants (like 's', 'ch', 'sh', 'f', 't', 'k') sound more or less harsh than others. Once I've found one that's too harsh, I often loop that consonant while looking at a high-resolution spectrum analyser and checking for spectral peaks that might be the source of the harshness. I'll then try cutting one or two frequencies with region-specific EQ to see if that helps. The same tactics work very well for spot-fixing sporadically harsh-sounding held notes — usually certain vowel sounds when sung loudly in the higher registers. Again, the high-midrange resonances that cause this are usually pretty easy to see on a spectrum analyser, once you've isolated the offending note.

Comparing the vocal mix balance on full-range speakers and a single-driver midrange speaker (I use my Avantone Mix Cube for this) can highlight where the upper or lower frequency extremes of a vocal are inconsistent. The trick is to get the vocal balance sounding consistent on the single-driver speaker first, then, if some notes feel too full/thin (or I think of it more as 'too big/small') on the full-range system, you can use EQ to rebalance the vocal's lower frequencies against its midrange. If some notes feel too bright/dull, or forward/recessed, the 'air' frequencies above 8kHz probably aren't consistent enough. The same comparison between speaker systems can be useful if trying to balance different vocal deliveries (eg. falsetto versus head voice, versus chest voice, versus belting...). If you get the different vocal deliveries sounding even on the small speaker but they then sound unstable on the big speakers, this implies that something about the vocal's frequency extremes isn't consistent with its midrange. One of the most common problems is where isolated falsetto notes sound well-balanced on the small speaker, but far too full/big once listening on your main monitors, because falsetto deliveries usually have significantly less true midrange energy.

Finally, high-pass filtering a vocal is particularly useful for pop/EDM, where you're trying to get that kind of consistently dense breathy intimacy. It focuses your ear on the level relationship between the consonants and vowels in that spectral range — the former are often much louder than the latter in project-studio recordings, but you'll also find differences in the HF energy and density of different vowels. I'll usually filter around 5kHz with a fairly steep slope (maybe 24dB/octave), but it depends on the voice.

SAMPLE LIBRARIES

Quiet Music
MagicDrum 2
Plug-in Instrument

★★★★★

Quiet Music's MagicDrum 2 is a multi-sampled steel tongue drum of around 1.7GB with on-board sound-shaping and processing that brings it into line with other recent Quiet Music offerings. For those unfamiliar with the tongue drum, its physical format is rather like an inverted steel bowl that has been cut so as to create a number of tuned tongues that can be played by striking with the fingers. These are generally designed to produce a specific musical scale rather than being chromatic. The sound is less dissonant than a steel drum but more complex than something like a Xylophone, giving it a haunting, musical quality that works very nicely with long reverbs.

Some 81 presets are available, most being a good fit for genres such as ambient, chillout and lo-f,i in addition to the more obvious relaxation-style music. The core sounds, many of which are already processed in interesting ways, can be further shaped via an ADSR envelope, low-pass or high-pass filter with adjustable cutoff, LFO modulation with a choice of waveforms, plus overall volume, pan, and tune. Modulation can also be controlled from a mod wheel, aftertouch or via velocity. The on-board effects comprise distortion, reverb, chorus and delay with tempo sync available for the delay times. These all have stomp-box style three-knob controls so no deep diving is required.

The sounds available from MagicDrum 2 range from pristine tongue drum sounds that respond naturally to playing velocity, to murky and mysterious versions of the core sound and on to sounds more reminiscent of temple gongs. Then there are patches such as Steel Sighs that almost totally disguise the source sound to produce something beautiful and synthetic. These sounds can be further customised via the on-board effect and controls, a simple-but-effective useful ploy being to add a slow attack to create bowed metal types of sounds.

For anyone working in the ambient or relaxation music genres, MagicDrum 2 is a simply fabulous instrument, especially given its low cost, the icing on the cake being the scale modes, in which a few moments of casual key tapping while exploring the various scales can produce unexpected and genuinely inspiring results. If you are still not sold on the idea, there's a free version with fewer scales and presets and if you already own the original MagicDrum, you get a preferential upgrade price. But that's enough talking about it, I'm going back to playing it!

Paul White

€36

www.quietmusic.eu

Pinkertone
Infinity Vol 1
Kontakt Instrument

★★★★★

Infinity Vol 1 is the first release from new sample library/virtual instrument development company Pinkertone. And, if you are a media composer working within the broad field of modern drama, action, horror or sci-fi, Infinity Vol 1 offers a palette of sounds that might well appeal. Essentially, the engine lets you blend up to four of the 660+ multi-sampled source sounds to create a whole range of drones, pulses, rhythmic beds, pitched impacts, sound effects and synthetic/hybrid playable sounds. No, it's not a revolutionary concept, but the neat front-end is nicely executed, offers plenty of sound design options, and easy enough to find your way around.

The 350+ supplied presets provide a good demonstration of the potential. These are divided into eight style-based categories; Drones, FX, High Tone Pulses, Reversers, Rhythms, Sequences, Synth and Tonal Hits. As suggested by the GUI, your preset can be created by blending any/all of the four sound slots, and each of these allows you to browse individual sounds from a dozen further categories such as drones, drums, effects, rhythms, impacts and sequences. There are some very cool individual sounds here with plenty of ammunition within the drone category and some great accent elements within the impact and tonal impact sections. Perhaps my only gripe is that it would be helpful to be able to audition sounds within the sound slot browsers prior to loading them. Maybe that's something for a future update?

The fairly minimalist design of the UI belies the impressive degree of sound control offered. Each sound slot provides an identical suite of controls accessed via the six icons beneath the slot. These offer mixer, envelope, step sequencing, effects (with four slots available), ambience and transpose options respectively. The central strip of the UI provides options for modulating the blend between the four sound sources. The Infinity Depth and Rate controls can automatically change the blend or, if preferred, you can disable this and assign the blend to a suitable MIDI CC number for manual control. The further controls provide similar modulation options for effects modulation. In short, there are a multitude of ways to let your inner sound designer do their thing.

In use, Infinity Vol 1 is a lot of fun. The presets are instantly inspiring, and this would make a great workhorse instrument for those underscore drama/tension cues that are a mainstay of modern TV and film soundtracks. Infinity Vol 1 is an impressive first product and busy media composers should certainly check out Pinkertone's demo videos on YouTube. I like this... When's Vol 2 due?

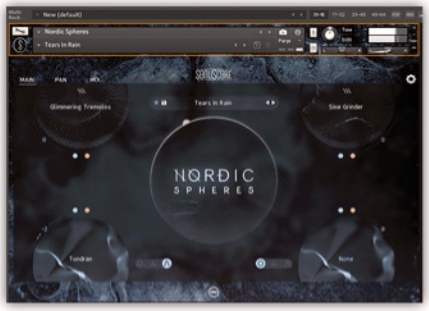
John Walden

$199

www.pinkertone.com

Sonuscore
Nordic Spheres
Kontakt Instrument

★★★★



If you check out John Walden's deep dive into Sonuscore's Time Textures for Kontakt (*SOS* December 2022), you'll find much in common with Nordic Spheres, insomuch as both focus on creating evolving sounds by allowing multiple parameters to be assigned to the central Macro wheel, which itself may be controlled from external MIDI sources, such as a mod wheel. Nordic Spheres, which works with both the full version of Kontakt and on the free Kontakt player, is inspired by the frozen lands of Iceland and combines carefully curated samples, categorised as Tonal, Non-tonal and Percussive. Granular processing can then be applied to two of the four layers enabling the instrument to produce of anything from assertive, pulsating atmospheres to foggy ambience and falling icicles. By layering sources and adding movement, the end result is a sonic feast of cinematic and ambient sounds. The layers each have their own ADSR and arpeggiator/LFO, and effects options that include EQ, saturation, chorus and tremolo. The sense of movement is provided by arpeggiator-style parameter control that can also be linked to the Macro knob.

The two upper layers (Tonal, Tonal Effects, Non-Tonal Effects, and Looping sounds) can utilise the Grain Engines to offer either smooth blurring or tinkly ear candy. The two more conventional sample engines handle Hits, Particles and Sustains. The GUI is uncluttered with the four layers shown on the Main view along with the large centre Macro wheel. Individual layers may be muted, solo'd or transposed in pitch. The slanting lines icon within each layer gets you to the more detailed instrument settings.

A Pan tab brings up the two pan engines, which can be set to any number of steps from three to 64 and can be tempo sync'ed. Mix reveals a four-channel mixer with controls for level, pan, delay send and reverb send, as well as a button to link the corresponding control to the motion engine. The outputs can be set to stereo or 5.1. Master effects comprise Compressor, Delay and Reverb.

The large selection of presets shows off the range of this instrument very effectively, though you can switch out samples very quickly to create your own variation before you decide to delve more deeply. I found many of the preset the sounds very inspirational, especially when experimenting with the movement aspect of the presets, and though few of the sounds would find a place in modern pop music, they are a perfect fit for TV/film soundtrack work and also a gift to anybody exploring the more adventurous side of ambient/chillout styles. Despite the title, not every preset will have you reaching for a warm jacket, but should you be asked to compose music to reflect cooler climes, you'll find plenty of screen-ready material here. *Paul White*

$199

www.sonuscore.com

Flame Sound
Firestarter
WAV

★★★★

For all the slick and sophisticated UI front-ends that allow you to manipulate the sounds within a sample-based library, in some sound categories there is still something to be said for the simple WAV file. That's a case you can certainly make for sound effects, whether that's Foley, ambience, spot effects or various types of impacts; just drag and drop a well-chosen WAV file onto the required spot on your timeline, adjust the level, and move on to the next task. It's not clever or complicated but it gets the job done. If that appeals, and you indulge in creating trailer sound, Flame Sound's Firestarter library might be just the thing. Available in two modestly priced versions — Standard and Deluxe — Firestarter provides a collection of trailer-friendly sound effects all served up in a well-organised set of categories and in a 48kHz/24-bit WAV format.

The two versions differ in the number of core SFX samples provided (125 vs 250) while the Deluxe version also bundles an additional 1GB of field recordings (street sounds, nature ambiences, etc) and a selection of soundscape samples for additional value. The samples supplied with both versions are categorised into the same nine folders — Accents, Booms, Braams, Downers, Impacts, Loops, Tension, Textures and Transitions — and the style of the contents in each is pretty much what you might expect. And, while that means that Flame Sound are not breaking dramatically new sonic ground here, the sounds themselves are consistently impressive.

That's particularly true of the various categories that contain sounds that offer spot effects designed to be placed as 'hit points' in the on-screen action such as the Accents, Booms, Braams, Downers and Impacts. Between them, they provide a good range of options. For example, the Accents go from fairly subtle bell and pluck sounds through to more weighty metal and synth tones. For bigger tasks, the Impacts category hits (doh!) the spot; there is plenty of weighty noise but without things getting too obviously hyped. The Booms bring a darker take on the same sort of thing if you need something a little more subtle with a low-end focus. I particularly liked the Braams. These come in three flavours — simple, impact and no impact — and sound really good.

If you are looking for a no-fuss collection of trailer-friendly sound effects to use either in a straight sound design role, or as ingredients to add into trailer-style music cues, Flamesound's Firestarter is a perfectly good place to start. The quality of the sounds is consistently high, and the simple WAV format makes it easy to drop and layer the sounds into any project regardless of the audio software platform in use. While similar palettes of SFX are offered by many other sample libraries, overall, Firestarter offers some very usable sounds at a very sensible price. *John Walden*

Standard $19, Deluxe $39.

www.flamesound.com

[Audio examples of this month's libraries are available at www.soundonsound.com.]

SOS SHOWCASE

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SOS WHY I LOVE...

THE NASHVILLE NUMBER SYSTEM

MIKE SENIOR

This time last year, I decided to pick up the bass guitar. Call it a mid-life crisis if you like, but after decades as a perma-pallid studio rat, I've recently emerged blinking into the limelight, accompanying assorted cover singers at a local weekly open-mic night. To aid me in this endeavour, I started building little lead sheets on my phone for some common song choices, but quickly ran into a couple of practical problems. Firstly, I couldn't fit an entire traditional lead sheet onto a single phone screen without the text becoming unreadably tiny — and scrolling while playing is unquestionably beyond my skill level! Secondly, many of the open-mic singers transpose the music to suit their vocal range, putting my carefully prepared lead sheets into the wrong key.

Which is why I now love the Nashville Number System (NNS)! This alternative method of charting songs notates chords as scale-degree numbers for the key in question, rather than with letter names. So, for instance, a "C Dm/F G C" chord sequence in C major translates into "1 2m/4 5 1". This immediately makes it much easier (with a little practice) to transpose songs into any required key. Now, I realise some people do something similar using Roman numerals — classical harmony textbooks are full of them, for instance. But is there honestly anyone who reads Roman numerals better than numbers? That is, anyone not wearing tweed? I don't know about you, but I'm forever mixing up IV with VI, even when I'm not sweating to keep pace with an ever-accelerating amateur drummer...

Besides, NNS also uses a type of rhythmic notation that's really space-efficient, because you don't need to write it on any kind of bars/beats grid, which is another huge part of the appeal for me. Each number is just assumed to be a full bar of that chord, although you can underline a group of numbers to indicate they're in the same bar and there are various other special symbols for indicating things like pushed/pulled chords, rests, and section repetitions. All of which means that, for example, the chorus of Adele's 'Easy On Me' becomes ":1..4/6 5..1/3 2m..>1/3 4:" and the full song chart ends up being genuinely tweetable. But I won't bore you with any more of the mechanics, because it's honestly super-easy to learn NNS from its Wikipedia page — as I did, in fact!

Beyond its manifest benefits in performance situations, however, I've also come to appreciate its value for comparing chord progressions from a songwriting/production perspective, because it makes the harmonic similarities between different songs immediately apparent, regardless of what keys those songs are in. So, for instance, did you know that the choruses of the Killers' 'Mr Brightside', Ed Sheeran's 'Castle On The Hill', and Walk The Moon's 'Shut Up And Dance' all share the same four-chord pattern, but with different harmonic rhythms? Or that Tracey Chapman's 'Fast Car' uses the same four-chord '1 6m 5 4' sequence as Prince's 'Purple Rain' and Spin Doctors' 'Two Princes', but starting on '4' rather than '1'?

So whether you're playing music or just stroking your chin about it, do yourself a favour and check out the Nashville Number system. If you're anything like me, you'll wonder what took you so long!

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